Softube

USER MANUAL

Supporting VST/VST3/AU/AAX Native and AAX DSP

Rev. Jan 31, 2020

intellijel

Summit Audio Inc.

Mutable Instruments

Solid State Logic
SOUND || VISION

OTO

ROLI

fix

Valley People

TUBE-TECH
LYDKRAFT DENMARK

TRIDENT
AUDIO
DEVELOPMENTS

DOEPFER
DRAWMER

Buchla

Tonelux®

Mutronics Limited

WEISS

Abbey
Road
Studios

EDEN
Bass Amplification

Marshall
AMPLIFICATION

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Softube User Manual

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Disclaimer

Every effort has been made to ensure that the information in this manual is accurate. However, there are a chance that we have made mistakes, and we hope that you understand that we are only humans. Please let us know about the mistake, and we'll fix it in the mix (or in the next version of this manual).

Support

On the Softube website (`www.softube.com`) you will find answers to common questions (FAQ) and other topics that might interest you.

Support questions can be posted at `http://www.softube.com`, where we will help you as fast as we can!

Web: `www.softube.com`
E-mail: `info@sof tube.com`
Phone: +46 13 21 1623 (9 am – 5 pm CET)

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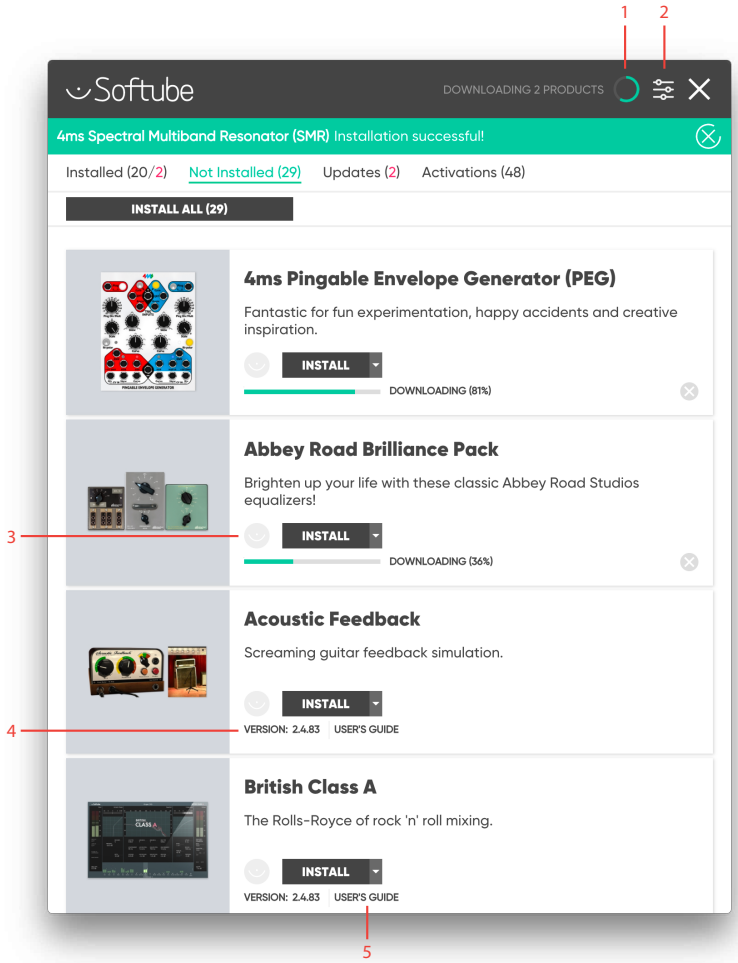
1 Softube Central

Welcome to Softube Central! Here you'll be able to manage installations and activations of Softube plug-ins. Softube Central is configured by default to manage the installations and activations automatically. If you have several workstations and want to manage it manually, you can activate the option in the Settings.

User interface

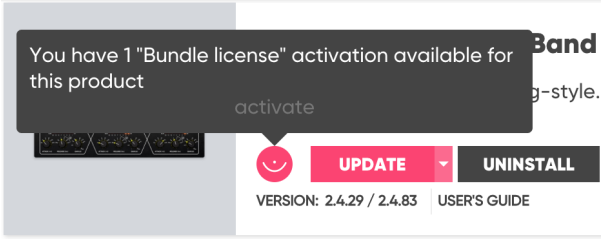


1. Status icon
2. Settings
3. Product status icon
4. Version
5. User's guide

Status icon
Status icon at the top right corner will inform you of the status of the Softube Central. In case it shows "Disconnected", please check your internet connection or restart the daemon application from the Softube Central settings.

Settings
Settings can be entered by clicking on the icon at the top right corner.

Product status icon
Hovering over the Product status icon will inform of the status of the available licenses.



Version
You can see the version of the plug-ins installed and the latest available version at the bottom of every product lane.

User's guide
You can access the user's guide from the product lane as well.

Note: for your convenience, Softube Central detects all available licenses on your ilok account automatically and will prompt you to uninstall products that you don't have licenses for.

Settings

Version shows the version of the Softube Central installed.

Account details section shows the registered email, user's name and the country.

PACE details section shows the status of the backend connection, the version of the daemon installer and PACE version used.

Current locations shows the ilok or machine based locations that are currently available.

Licenses section includes a switch with **Auto Activate** mode. If set to Auto mode, the Softube Central will activate the licenses automatically after plug-in installation. If set to Manual mode, the Activations tab will appear where you'll be able to manage the license activations manually.

The Activations tab will show the available activations and locations where the license was activated.

Note: in case the location is unavailable the Softube Central will grey out the location and mark it as unavailable. If it shows the ilok location as unavailable with ilok plugged in, please check if the latest ILOK drivers are installed.

Softube Central in use

After Softube Central installation please launch it from the Applications folder on your Mac computer. It will start automatically on PC.

Please follow the instructions on additional installations the Softube Central requires. When the installations are completed, go to Not Installed tab and click Install All button, Softube Central will install all purchased plug-ins and the plug-ins with active 20-day DEMO activations.

Softube Central will install and activate the plug-ins for you. Now you can close the Softube Central and make some music!

Note: it's helpful to run Softube Central once in a while to check if any updates are available.

Note: Softube Central public beta version doesn't have the update license functionality yet. We are working on it. Please handle the license update through the ilok manager.

System Requirements

These are the system requirements for all Softube plug-ins. **Please check the latest requirements on www.softube.com.**

All native versions

- Mac OS X 10.12 or newer
- Windows 64-bit, versions 7, 8 or 10
- Please Note: Softube plug-ins support 64-bit hosts only, and a 64-bit OS is required. Therefore, for example, Pro Tools 10.x is no longer supported (contact support for latest 32-bit compatible instalers).
- Intel Core 2 Duo, AMD Athlon 64 X2 or newer
- Screen resolution larger than 1280x800
- 8 GB RAM or more is recommended, as well as at least 8 GB hard disk space for installation (individual plug-ins take less space, while sample libraries may require additional disk space. Specific information available on each instrument's page).
- Softube account
- iLok account
- Internet access for downloading the plug-in (s) and managing the license(s)
- iLok generation 2 and up, if you intend to use an iLok dongle for authorizing your plug-ins (note that an iLok dongle is not a requirement)
- In theory, any 64-bit VST, VST3, AU, or AAX (11.0.2 or higher) compatible host application should work. However, due to plug-in host differences between DAWs - and our own rigorous testing standards - we only officially test our plug-ins and instruments in the most recent versions of Pro Tools, Logic Pro, Cubase, Live, Studio One, Digital Performer, Reaper and Sonar. Softube plugins are not

tested in the non-listed systems, but they most likely work as long as the system requirements are met. We cannot guarantee a solution for issues in unsupported systems.

- Supported sample rates: 44.1, 48, 88.2, 96, 176.4 and 192 kHz, in both mono and stereo.
- The most recent maintenance release of the DAW application is recommended.

Please Note: Pro Tools 10 is not officially qualified by Avid higher than OS X 10.8

AAX DSP version

For AAX DSP you need to meet the above requirements for the Native versions. You also need to have an Avid HDX processing card. For plug-ins available for AAX DSP, see Which Softube plug-ins support AAX DSP?

AAX DSP Supported Sample Rates

The following sample rates are currently supported for the AAX DSP format.

	44.1/48 kHz	88.2/96 kHz	176.4/192 kHz
Active Equalizer	√	√	√ (mono)
Acoustic Feedback	√ (mono)	√ (mono)	√ (mono)
Acoustic Feedback Deluxe			
Bass Amp Room	√	√	√ (mono)
FET Compressor	√	√	√
Focusing Equalizer	√	√	√
Metal Amp Room	√	√	
Passive Equalizer	√	√	√
Spring Reverb	√	√	√
Summit Audio Grand Channel	√	√	√
Summit Audio EQF-100	√	√	√
Summit Audio TLA-100A	√	√	√
Tonelux Tilt	√	√	√
Tonelux Tilt Live	√	√	√
Trident A-Range	√	√	√
TSAR-1 & TSAR-1R Reverb			
Tube Delay	√	√	√ (mono)
Tube-Tech Classic Channel	√	√	√
Tube-Tech CL 1B	√	√	√
Tube-Tech ME 1B	√	√	√
Tube-Tech PE 1C	√	√	√
Valley People Dyna-mite	√	√	√
Vintage Amp Room	√	√	√ (mono)
White Amp	√	√	√ (mono)

√ = sample rate supported in both mono and stereo.
√ (mono) = sample rate only supported in mono. Use “multi-mono” for stereo.

Plug-In Latencies

This table shows the plug-in latency in samples. These numbers are also reported to the DAW.

	44.1/48 kHz	88.2/96 kHz	176.4/192 kHz
Active Equalizer	8 samples	0 (no latency)	0 (no latency)
Acoustic Feedback	0	0	0
Acoustic Feedback Deluxe	0	0	0
Bass Amp Room	3	0	0
Drawmer 1973	4	0	0
Drawmer S73	4	0	0
FET Compressor	4	0	0
Fix Flanger	2650/2884	5292/5760	10584/11520
Fix Doubler	4	0	0
Focusing Equalizer	4	0	0
Metal Amp Room	3	0	0
Modular	2	2	0
Mutator	3	0	0
Passive Equalizer	8	0	0
Spring Reverb	0	0	0
Summit Audio Grand Channel	8	4	0
Summit Audio EQF-100	4	0	0
Summit Audio TLA-100A	4	4	0
Tonelux Tilt	0	0	0
Tonelux Tilt Live	0	0	0
Transient Shaper	8	8/9	16/17
Trident A-Range	4	0	0
TSAR-1 & TSAR-1R Reverb	0	0	0
Tube Delay	3	0	0

	44.1/48 kHz	88.2/96 kHz	176.4/192 kHz
Tube-Tech Classic Channel	4	4	0
Tube-Tech CL 1B	4	0	0
Tube-Tech ME 1B	12	8	0
Tube-Tech PE 1C	12	8	0
Spring Reverb	0	0	0
Summit Audio TLA-100A	4	4	0
Valley People Dyna-mite	4	0	0
Vintage Amp Room	3	0	0

2 User Interface

SOFTUBE PLUG-INS ARE "what you see is what you get" products. You should be able to intuitively learn the products within minutes, so that you can work fast and efficient with them. There are a couple of things that remain the same for all of our plug-ins, like keyboard commands and menu row. These will be explained in this chapter. For detailed information of a particular plug-in, please see its chapter.

Menu Row

In the bottom of the plug-in interface, you will see a thin black row with some buttons. We'll use the Dyna-mite plug-in as example, but the same goes for all plug-ins.

1. "About" box with version information
2. Value display
3. Setup
4. Open quick guide (if applicable)
5. Open manual in PDF reader

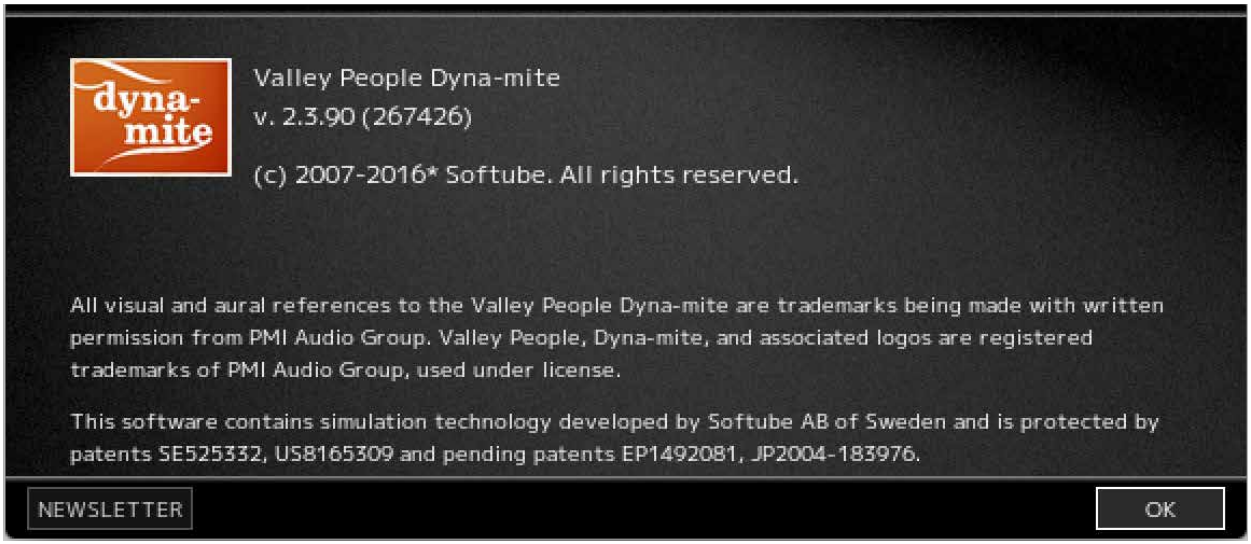
About Box Open the "About" Box with version info.

Value Display Displays the knob value when the mouse is pointing at the knob.

Setup Changes global options for all instances of that plug-in.

Quick Guide Opens the Quick Guide (if applicable) in a PDF reader.

Open Manual Opens the User Manual in a PDF reader.


dyna-mite
Valley People Dyna-mite
v. 2.3.90 (267426)
(c) 2007-2016* Softube. All rights reserved.
All visual and aural references to the Valley People Dyna-mite are trademarks being made with written permission from PMI Audio Group. Valley People, Dyna-mite, and associated logos are registered trademarks of PMI Audio Group, used under license.
This software contains simulation technology developed by Softube AB of Sweden and is protected by patents SE525332, US8165309 and pending patents EP1492081, JP2004-183976.
NEWSLETTER
OK

About Box

Clicking on the Softube logo opens the "About" box, with information about the current version, trademarks and legal yada-yada. That's not really interesting, but the cool stuff is the buttons on the bottom row.

Newsletter	Opens the web browser and lets you sign up for our newsletter, so that you will be the first to know about new products, tips & tricks, updated versions, etc.
OK	Closes the "About" box.

Setup

In the Setup window you can change settings that will affect all instances of that particular plug-in. If you for example de-select the "Show Value Display" option in the Valley People Dyna-mite plug-in the value display will be off for all Dyna-mites on your system until you select that option again.

The different options vary between Windows and Mac, and also different formats and plug-ins. The most common options are:

ENABLE EXTERNAL SIDECHAIN SUPPORT: If you want to use external sidechaining in VST2.

SHOW VALUE DISPLAY: Enables the parameter and value display in the bottom row of the plug-in.

PREFER COCOA GUI: 32-bit Mac Only. Tries to use a newer graphics system. Select this if you experience problems with the user interface.

> You need to restart your host software (DAW) before the changes to fully take effect!

If you messed something up and manually need to set these options, you'll find them in text format in the following locations:

MAC OS: `~/Library/Application Support/Softube`

WINDOWS: `username\Application Data\`

Key Commands

All numbers and labels in the plug-in are clickable. This allows you to easy select a setting by clicking on the wanted value. Hovering above a label will turn the mouse pointer into a pointing hand.

Mouse

Up/Down or Mouse Wheel	Change a parameter, such as a knob or a switch.

Keyboard

Fine Adjust	⌘ (Mac) or CTRL (Win), while changing the parameter value.
Reset to Default	ALT, while clicking on the knob or fader.

Pro Tools Specific Commands

Automation Control Window	CTRL+ALT+WIN+click (Win) or CTRL+⌘+ALT+click (Mac)
Show Automation Track	CTRL+WIN+click (Win) or CTRL+⌘+click (Mac)

Plug-In Specific Key Commands

In many plug-ins, you can SHIFT-click on a knob or a switch to get some extra functionality. Below are some examples. See the chapter for each product for more information.

Metal Amp Room

SHIFT-click and drag a mic will move both mics simultaneously.

All Amp Rooms

SHIFT-click in the cabinet background will change cabinet (or amp) without any animations

Focusing Equalizer

SHIFT-click on the **Low** and **High Cut** faders will move both faders simultaneously.

Modular

In Modular, hold SHIFT and click to add several modules at once to the rack.

Mutronics Mutator

SHIFT-click on a parameter changes that parameter for both the left and right channels.

Drawmer 1973 Multi-Band Compressor

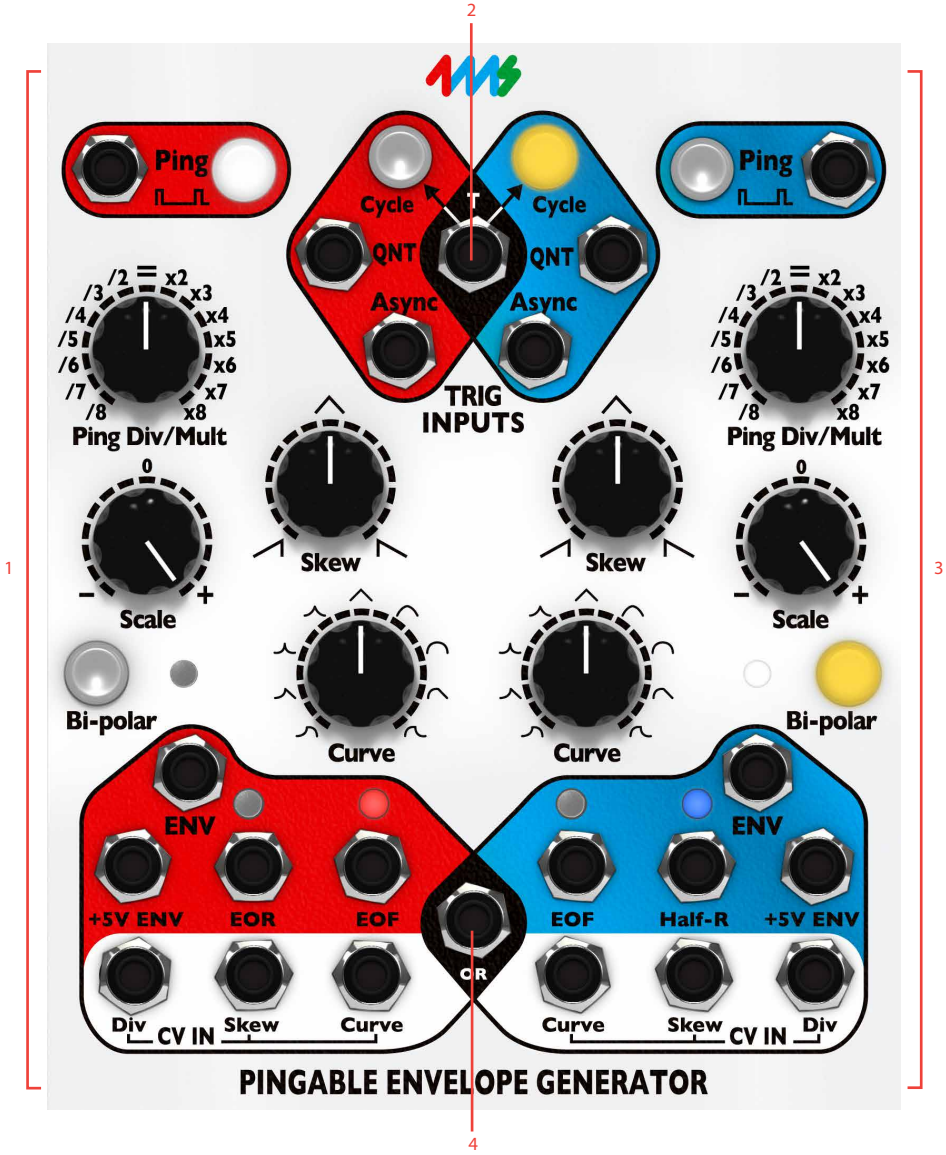
SHIFT-click on a parameter changes that parameter for all three bands. SHIFT-clicking the **Status** label will solo that band.

MODULAR EXCLUSIVE

3 4ms Pingable Envelope Generator (PEG)

The 4ms Pingable Envelope Generator (PEG) is a dual envelope generator whose envelope lengths are set by the time between external clock pulses or "pings". The dual envelopes are of simple two-stage, rise-and-fall design, and can be used in cycle mode as LFOs. When gated it can also serve as a simple rise-sustain-fall (or ASR) envelope, commonly used in monophonic synths of the 70s.

Overview



1. Red channel
2. Toggle for both channels
3. Blue channel
4. Combined output

Description

PEG consists of two identical envelope channels, Red and Blue. They can be used independently or via the mixed (OR) output.

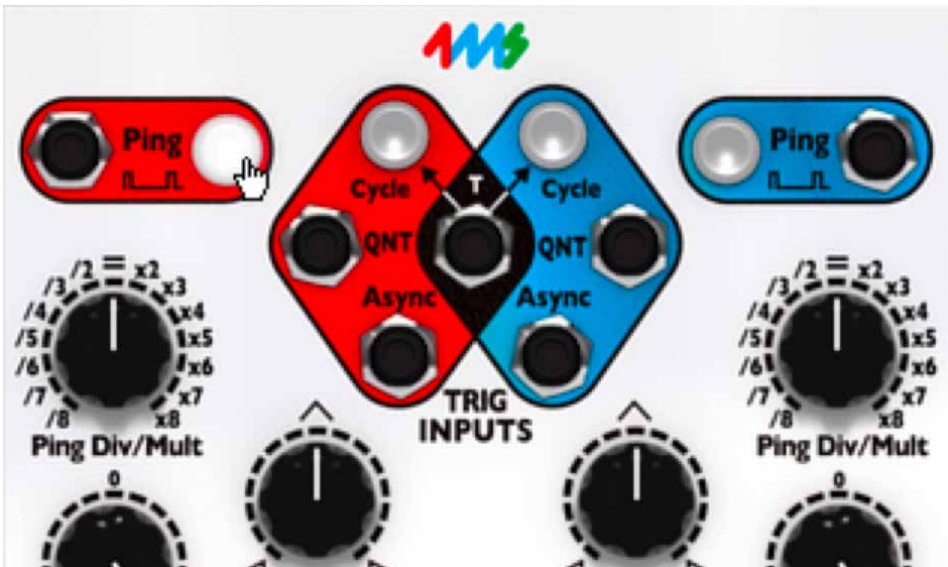
In PEG, CV control over envelope shape, skew and ping (clock) division/multiplication per envelope channel (Red/Blue) is possible. PEG features a multitude of different triggering and cycling options (AD, AR, quantization, cycle, cycle toggle) as well as different level stage trigger outputs (EOR, EOF and Half-R).

Each of the envelope channels also features a Tap tempo button for manual pinging.

The firmware features used in the PEG firmware version 4.3 are all emulated by Softube except the 2 second tap clear feature and System Mode (factory presets are being used).

Getting Started With PEG

1. Set the tempo of an envelope channel by repeatedly clicking on one of the ping buttons at the top of the PEG module. "Pinging" an envelope channel will set the internal tempo for that envelope and determine its overall length. An envelope channel can also be "pinged" by patching an external clock into its Ping input jack.



2. Use the pinged envelope channel as a rise-sustain-fall (same as an ordinary ASR) envelope by patching a gate into the Async jack. Now, when a gate appears on the Async jack, the rise-portion of the envelope will play through and the envelope will then stay at its maximum level (i e sustain level) until the gate is released. When the gate is released, the fall-portion of the envelope will be initiated and played through.

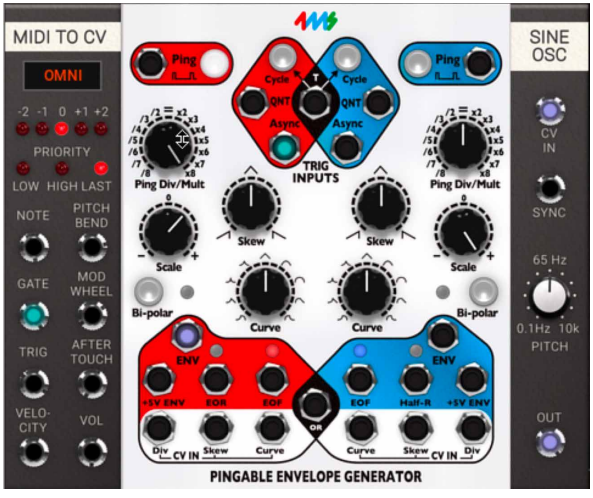
---

3. PEG does not normally produce sound on its own, so patch the PEG Envelope CV output to the whatever module you want to modulate. This could be the pitch of a oscillator, the cutoff frequency of a filter or a voltage controlled amplifier (VCA), just to mention the most common use-cases.

4. The output envelope amount can be adjusted by using the Scale knob. It can also be used to invert the output envelope CV.

---

5. Experiment with different envelope lengths by "pinging" the envelope at different tempos. You can also change the speed by turning the Ping Div/Mult knob to set the envelope to be up to 8 times faster or slower then pinged tempo. Changing the Curve or Skew of the envelope will not affect the overall rise and fall time, only the balance between the two.



6. Use the cycle function by clicking on the Cycle button at the top if you want the envelope to keep on repeating independently. This is great when you want to use PEG as an LFO.

Parameters

Red Tap Button

This is the manual tap tempo button for the Red envelope channel. Manually "ping" the red envelope channel to a certain tempo by repeatedly clicking on this button. The button will flash in sync with tapped tempo.

Note: Pinged PEG tempos are saved in a Modular patch, but the phase difference between the two envelope channels (or other PEG modules) will not be saved. This is to avoid other (in our opinion worse) side effects of saving the phase and this can make patches using cycle sound different when loaded than compared to when they were saved.

Red Cycle Button

This turns Red envelope cycle on and off. When cycle is enabled the envelope will keep on triggering on every ping.

Blue Cycle Button

This turns Blue envelope cycle on and off. When cycle is enabled the envelope will keep on triggering on every ping.

Blue Tap Button

The manual tap tempo button for the Blue envelope channel. "Ping" the red envelope channel manually in a certain tempo by clicking on this button. The button will flash in sync with tapped tempo. If more than two clicks are received, the average time between the three latest clicks will set the pace of the envelope (unless the third click occurs more than 50% different than the timing period of the first two clicks).

Red Ping Div/Mult Knob

The Div/Mult knob sets the offset division or multiplication of the pinged tempo of the Envelope. If this knob is set to "=" (preset value), no division or multiplication takes place. Division makes the output envelope times longer (up to 8 times longer) and Multiplication makes

the output envelope times shorter (down to a 1/8 of pinged timed).

Blue Ping Div/Mult Knob
The Division / Multiplication offset for the Blue envelope generator (see description above).

Red Skew Knob
The Red Skew knob sets the proportion between the rise and fall times of the Red envelope generator. All the way counter-clockwise the rise portion is the dominant (longer) part. With the Skew knob set at 12 o'clock both portions are equal and with the Skew knob set all the way clockwise the fall portion becomes the dominant (longer) part.

Red Skew Knob
The Blue Skew knob sets the proportion between the rise and fall times of the Blue envelope generator in a similar manner as the Red Skew knob (see description above).

Red Scale Knob
This knob sets the envelope amount and polarity output at the ENV (red) output jack. With the Red Scale knob set all the way counter-clockwise, the ENV (red) output will output a fully inverted (negative) envelope shape (-10V), while the knob set at 12 o'clock (0), will produce no envelope output at the ENV (red) output jack. Fully clockwise (preset position) will produce a full positive envelope output (+10V) at the ENV (red) output. The +5V ENV (red) output jack is not affected by the Red Scale knob and will always output a fully positive envelope shape.

Blue Scale Knob
The Blue Scale knob has a similar behavior as the Red Scale knob but will of course affect the ENV (blue) output jack instead (see description of the Red Scale knob above). The +5V ENV (blue) output jack is not affected by the Blue Scale knob and will always output a fully positive envelope shape.

Red Bi-Polar Button
When engaged, this button centers the Red envelope around 0V with a approximate span of -5V to +5V.

Red Curve Selection
This knob selects the desired envelope-curve for the red envelope from a plethora of 17 different exponential, linear and logarithmic waveforms. As you turn the knob clockwise the first 4 curves are asymmetrical with exponential attacks and different decays, while the next 4 curves are symmetrical, ranging from exponential to logarithmic in 4 interpolated steps. Middle curve is linear (triangle wave). Curves 10 to 13 are symmetrical, ranging from linear to logarithmic, and the last 4 curves are asymmetrical with logarithmic attacks and varying decays (logarithmic to exponential).

Blue Curve Selection
This knob selects the desired envelope curve for the blue envelope with a similar function (see description above).

Blue Bi-Polar Button
This button centers the Blue envelope around 0V with an approximate span of -5V to +5V.

Tip: Use envelope Cycle mode if you want a PEG channel to work as an LFO.

Indicators

Red Ping LED
This LED button indicates the pinged tempo of the Red Envelope with tempo alterations set by the Red Div/Mult knob.

Red Cycle button LED
This LED button indicates whether or not the Red Envelope is in cycle mode or not (see further description above).

Blue Cycle button LED
Indicates whether or not the Blue Envelope is in cycle mode or not (see further description above).

Blue Ping LED
Indicates the pinged tempo of the Blue Envelope with tempo alterations set by the Blue Div/Mult knob.

Red Bi-Polar button LED
This LED button indicates if Bi-Polar mode engaged for the Red Envelope or not.

Red Envelope LED
This LED lamp reflects the output of the +5V ENV (red) jack and thus does not reflect amount or polarity of the ENV (red) output.

Blue Envelope LED
This LED lamp reflects the output of the +5V ENV (blue) jack and thus does not reflect amount or polarity of the ENV (blue) output.

Blue Bi-Polar button LED
indicates if Bi-Polar mode is engaged for the Blue Envelope or not.

Red EOR LED
When the red envelope finishes the rise portion of its cycle, the EOR (red) output goes high (+5V) and this LED is lit.

Red EOF LED
When the red envelope finishes the fall portion of its cycle, the EOF (red) output goes high (+5V) and this LED is lit.

Blue EOF LED
When the blue envelope finishes the fall portion of its cycle, the EOF (blue) output goes high (+5V) and this LED is lit.

Blue Half-R LED
The Blue Half-R LED is lit when the blue envelope has gone through half (50%) of its rise portion.

Inputs

Ping (Red) Input
Short pulses sent into this input "ping" the red envelope channel and set the internal tempo for it. Faster pulses sent here means shorter envelope, and pulses further apart will effectively result in a slower envelope. Tempo is derived from the time between the two latest pulses.

Ping (Blue) Input
Works similar to the ping (red) input, but obviously affects only the blue envelope.

QNT (Red) Input
This is the quantized input for the red envelope. Quantized means that any pulse that appears here will not trigger the envelope until the next "ping" is indicated on the red envelope channel LED. If gated (this input held high, at +5V) for a period longer than two pings in succession, the envelope will repeat (similar effect to cycle mode).

T (Toggle) Input
When a gate is applied (this input held high, at +5V)

to this input, both envelope channels' Cycle buttons will toggle state, on to off and vice versa. When the gate is released they will revert to their previous state. The "T" jack is useful for toggling between the two channels: set one channel in Cycle mode and the other channel to non-cycling while using the output from the OR jack (see the outputs section below). The "T" jack is also useful for turning both channels on/off at the same time.

QNT (Blue) Input
Blue envelope channel quantized input. Works similar to the QNT (red) input described above but for the blue envelope channel.

Async (Red) Input
When a trigger is received at this jack, the red envelope will trigger or re-trigger immediately. Holding a high gate (+5V) on this jack causes a sustain period followed by a fall period when the gate goes low (ASR envelope).

Async (Blue) Input
Work similar to the Async (red) input jack.

Div (Red) CV Input
A CV signal patched to this input will add to the value set by the Red Ping Div/Mult knob. Range is -5V to +5V.

Skew (Red) CV Input
CV control over the Red envelope Skew. This will add to the value set by the Red Skew knob. Range is -5V to +5V.

Curve (Red) CV Input
CV control over the Red envelope Curve. This will add to the value set by the Red Curve knob. Range is -5V to +5V.

Div (Blue) CV Input
A CV signal patched to this input will add to the value set by the Blue Ping Div/Mult knob. Range is -5V to +5V.

Skew (Blue) CV Input
CV control over the Blue envelope Skew. This will add to the value set by the Blue Skew knob. Range is -5V to +5V.

Curve (Blue) CV Input
CV control over the Blue envelope Curve. This will add to the value set by the Red Curve knob. Range is -5V to +5V.

Outputs

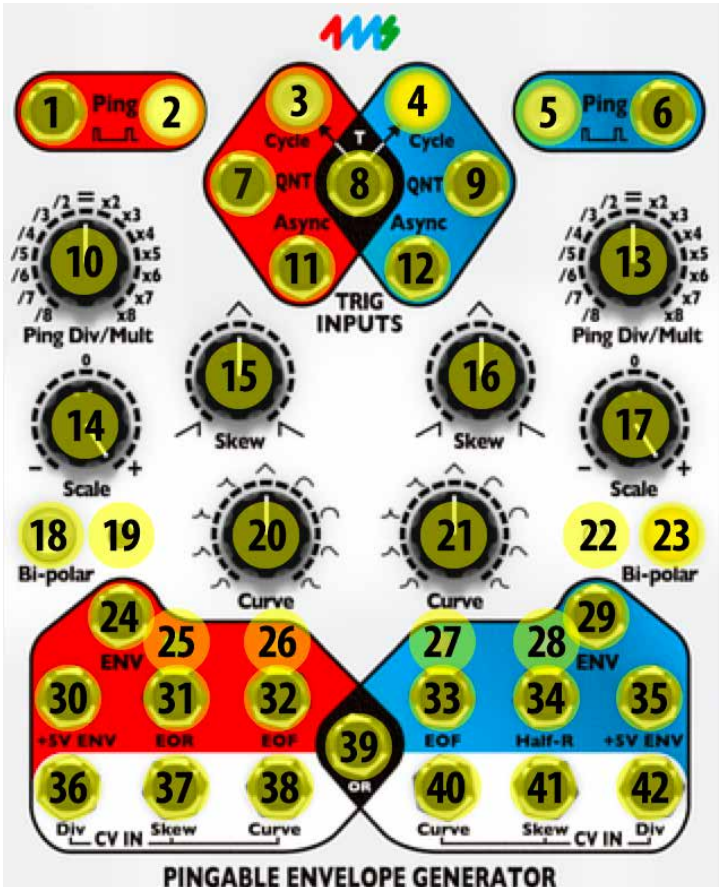
ENV (Red) Output
This is the main envelope CV output of the red envelope channel. The CV amount output here is controlled by the Red Scale knob and Red Bi-Polar button. Range is -10V to +10V.

ENV (Blue) Output
This is the main envelope CV output of the blue envelope channel. The CV amount output here is controlled by the Blue Scale knob and Blue Bi-Polar button. Range is -10V to +10V.

+5V ENV (Red) Output
This is the unregulated envelope CV output of the red envelope channel. Range is always 0V to +5V.

EOR (Red) Output
This jack output +5V at the end of the rise portion of the envelope when the fall portion is initiated. Note that this jack does not output +5V during sustain when gating the Async input jack.

EOF (Red) Output
This jack outputs +5V at the end of the fall portion of the red envelope or when the rise-portion is in progress.



Parameters, Indicators, Inputs and Outputs

1. Ping (Red) Input
2. Red Tap Button
3. Red Cycle button LED
4. Blue Cycle button LED
5. Blue Tap Button
6. Ping (Blue) Input
7. QNT (Red) Input
8. T (Toggle) Input
9. QNT (Blue) Input
10. Red Ping Div/Mult Knob
11. Async (Red) Input
12. Async (Blue) input
13. Blue Ping Div/Mult Knob
14. Red Scale Knob
15. Red Skew Knob
16. Blue Skew Knob
17. Blue Scale Knob
18. Red Bi-Polar Button
19. Red Bi-Polar Button LED
20. Red Curve Selection
21. Blue Curve Selection
22. Blue Bi-Polar Button LED
23. Blue Bi-Polar Button
24. ENV (Red) Output
25. Red EOR LED
26. Red EOF LED
27. Blue EOF LED
28. Blue Half-R LED
29. ENV (Blue) Output
30. +5V ENV (Red) Output
31. EOR (Red) Output
32. EOF (Red) Output
33. EOF (Blue) Output
34. Half-R (Blue) Output
35. +5V ENV (Blue) Output
36. Div (Red) CV Input
37. Skew (Red) CV Input
38. Curve (Red) CV Input
39. OR (Black) Output
40. Curve (Blue) CV Input
41. Skew (Blue) CV Input
42. Div (Blue) CV Input

EOF (Blue) Output

This jack outputs +5V at the end of the fall portion of the blue envelope or when the rise-portion is in progress.

Half-R (Blue) Output

This jack outputs +5V when half of the rise portion of the blue envelope has been played through. Note that this jack also output +5V during sustain when gating the Async input jack.

+5V Env (Blue) Output

This is the unregulated envelope CV output of the blue envelope channel. Range is always 0V to +5V.

OR (Black) Output

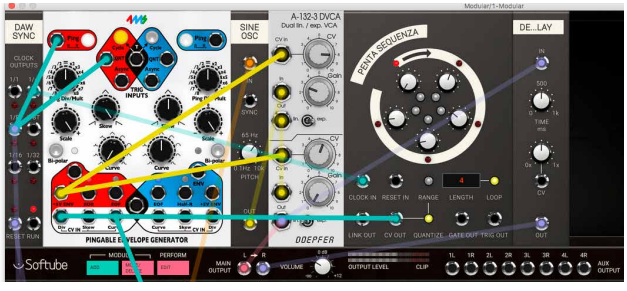
The OR jack will output the highest voltage value from either the red or blue ENV jack at any given moment. One way to use this is to think of the OR jack as a mix out, and use the Scale knobs as level knobs and the Bi-Polar buttons to bring down the relative level of a channel (kind of like a mute button).

The PEG Module In Use

The 4ms Pingable Envelope Generator is a very versatile module with many fun uses. Here are some suggestions:

1. CV controllable length of the PEG cycle - Use a CV-controllable LFO or VCO to ping the tempo of PEG. By varying the pitch of the driver the length of the PEG envelope can then be easily be altered, for example via an additional LFO or sequencer.

2. Ratcheting envelopes - Clock one envelope module from the DAW sync clock module with a steady pace, for example quarter notes. Use another faster input to clock the QNT input while an LFO is sweeping the Div input. The pace of the faster input, for example 32nd notes, will set the upper limit to the ratcheting speed. If you desire a more controlled approach to ratcheting, using a CV to turn on and off the cycle function is the way to go.

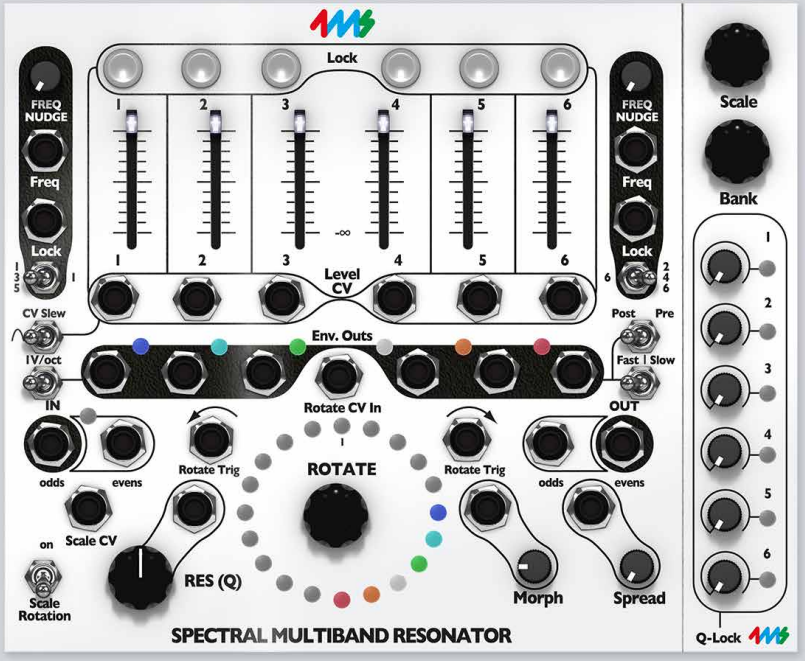


3. DAW synced LFOs - Use the DAW sync module to ping a cycling PEG at a regular pace, for example by using the 16th notes output. The reset jack or whole note jack should also be connected to the Async input to ensure PEG to start at the same phase each time (or each bar when using the whole note output). Using the Tri (preset) shape will be the most common use case when using PEG as an LFO, but of course you can use any shape to create your own customized LFO.

4. PEG Percussion - PEG is a great module to use for percussive sounds. Use the Skew and Curve knobs to shape the pitch bend of the tuned percussion and the Scale knob to set pitch bend amount. Experiment with different curve types for different sounds.

Credits

Oscar Öberg – modeling, project management, validation. **Kristofer Ulfves** – presets, validation, user manual. **Igor Miná** – user manual layout, hardware photos. **Niklas Odelholm** – GUI graphics.

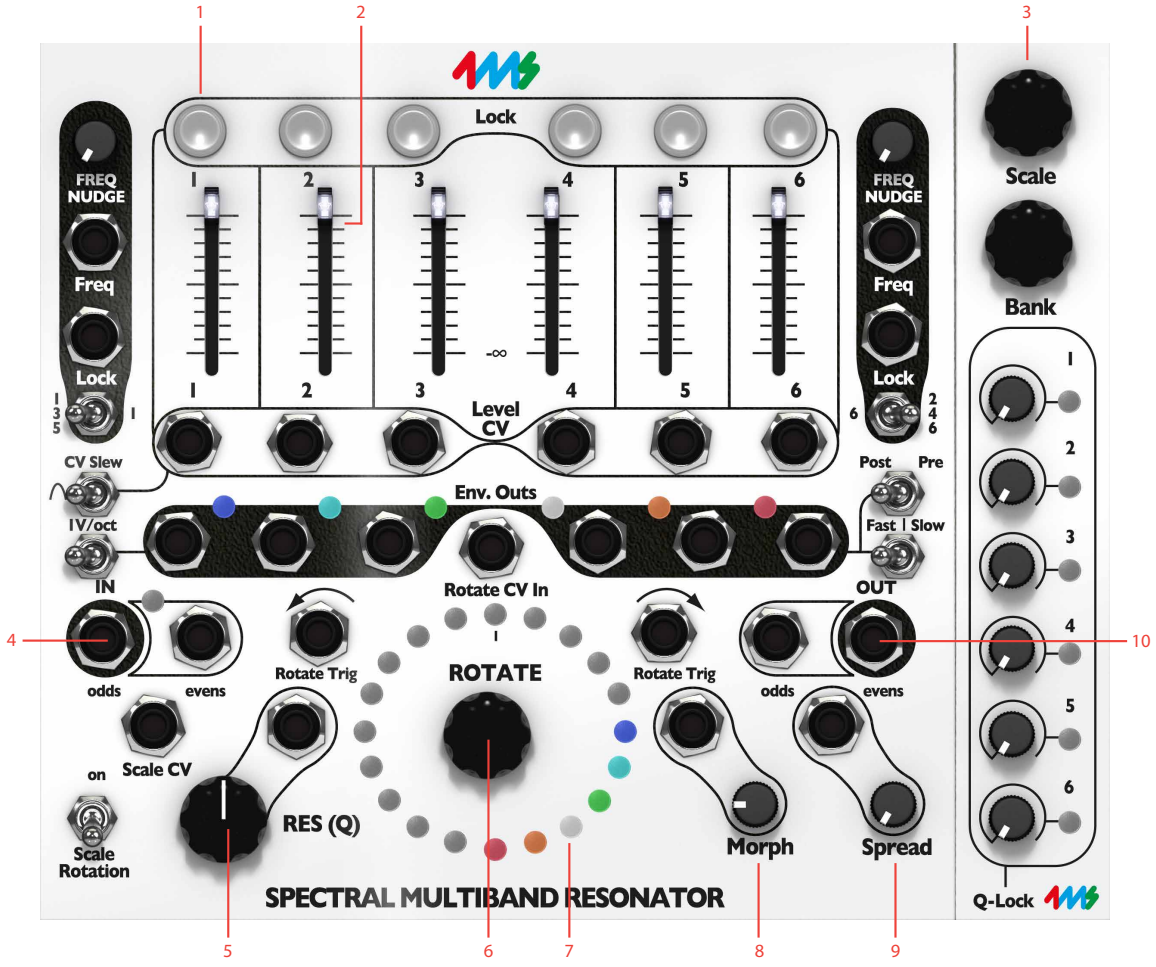


MODULAR EXCLUSIVE

4 4ms Spectral Multiband Resonator (SMR)

THE SPECTRAL MULTIBAND RESONATOR FROM **4ms** is an innovative multiband resonant filter which can process audio like a classic filter bank, ring like a marimba when plucked/struck, vocode, re-mix tracks, harmonize, output spectral data, quantize audio to scales and much more.

The SMR consists of 6 channels of resonant bandpass filters, all individually locked to a 20 notes scale displayed in the centre of the unit via the LED ring. All 6 filters can easily be swept around their present note scale manually or via CV. Notes scales cover everything from western scales, alternative tunings to more esoteric *Gamelan* and *microtonal* scales. Each filter's output volume can be controlled via CV and each



1. Filter band Lock buttons
2. Filter band Volume sliders
3. Expander section
4. Inputs
5. Resonance
6. Rotate
7. LED ring
8. Morph
9. Spread
10. Outputs

channel also has an envelope follower that can be set to output different kinds of modulation and pitch CV.

The official firmware v4 of the original 4ms hardware was mainly used when modeling the SMR behavior, the exception being the added optional 1 V/oct switch functionality of firmware v5 for the envelope follower outputs (see detailed description below) and the lack of user definable scales. The experimental non-tracking filter type is also not featured in this version.

User Interface

The interface might look daunting at a first glance, but is not that difficult. The most important dials, buttons and jacks are explained in the overview below.

Color Schemes

The SMR consist of 6 bandpass filters. Each filter-band is identified by a number and a color, which is shown on the LED ring. Each filter can be set to almost any frequency, except that it has to be within one of the available scales. More on that below.

Filter-band 1 is always deep blue, band 2 cyan, band 3 green, band 4 white, band 5 orange and band 6 red. This color reference is also shown on each filter-band's Env Out LED indicator.

The LED ring has a second function: It switches function temporarily when operating the scale and bank knobs, giving information about the scales' relation to the different filter-channels.

Description of Banks and Scales

The SMR is all about scale rotation. The LED ring is an important indicator to watch in order to understand the state of each of the different filter channels in relation to scale rotation. Rotation can be either clockwise or counter-clockwise.

When rotation occurs, all six bands will move in the same direction and the six lights on the light ring will fade to the next available free spot. If a band is locked it will not rotate, and that band will instead be indicated by a blinking light corresponding to that band in the LED ring. If a band is rotated such that it is intent to land on top of a locked band or on another band moving in the opposite direction (which occurs when spread is increased), it will then move past the locked or moving band to the next available free spot on the other side.

Rotation is closely linked to **Morph**, which sets the pace at which rotation can happen. No matter how much rotation you tell the SMR to do, it will only go as fast as the Morph setting allows. Rotation is also queued, so if you quickly turn the **Rotate** encoder ten notches to the right and have a very slow Morph setting, the SMR will start rotating all the channels one step at a time until it's rotated a total of ten times. Each of the ten rotations will happen at the Morph speed. Reversing direction will clear the queue, so if you want the SMR to stop rotating, you can turn the **Rotate** encoder one click in the other direction. The SMR will forget the remaining rotations in the queue and just rotate once in the new direction.

Getting started with SMR

Here are a couple of guides to quickly get you started with the SMR.

Tutorial 1: Droning

Preset patch "**Exp SMR tutorial 1**"

The easiest SMR patch is making droning chords. Only one cable is needed! Set up the SMR as follows:

1. Add the SMR module to your empty rack by clicking on the **Add** button in modular.
2. Connect a cable from the **Evens** output jack (marked black on panel) to you main output.
3. Turn up the **Res (Q)** knob to 100%.
4. At the top of the SMR are the six **Volume** sliders. Each one controls the volume of a filter/resonator channel. Push down all six sliders, then slide each one up, one at a time. You should hear a pitch fade up in volume. Each slider controls a different frequency.
5. As you adjust the slider, watch the LED ring. Each slider is associated with a color. The color gets brighter on the LED ring as the slider moves up and down. Each of the 20 spots on the LED ring is associated with a pitch (a frequency, or a musical note).
6. Now turn the **Rotate** encoder clockwise by clicking and dragging upwards. Hear how the pitches shift up and the lights rotate one step clockwise. Click and drag the **Rotate** encoder downwards so it rotates counter clockwise. Hear how the pitches shift down and see how the lights rotate counter-clockwise.
7. Keep on spinning the **Rotate** encoder so that the lights move past 12:00 (due north, marked by a tick mark on the light ring). Now notice how the pitches go to the top and then start at the bottom. By rotating and pushing different sliders up and down, you can make different chords.

Connect the output of the SMR to the Main Output of Modular.

8. Play with the **Morph** knob as you continue to turn the **Rotate** encoder knob. **Morph** will set the speed at which rotation happens. With **Morph** at 100%, when you turn the **Rotate** encoder, the SMR will slowly fade from one position to the next. With **Morph** at 0%, the SMR will instantly jump from one position to the next (this can sound "clicky").
9. Now try playing with the **Spread** knob, which controls the number of empty positions in between each channel in the scale. With **Spread** at 0%, the channels occupy adjacent spots on the LED ring. As you turn **Spread** knob up slowly, the channels will jump from having one position between them, then two positions, then three and so on. As the spacing increases, the channels will get pushed around and eventually the highest channel will wrap around the lowest.
10. Try out playing with the **Q (Res)** knob. The SMR has a digital noise source normalized to the audio IN jacks. With the **Q (Res)** knob set to 0%, you should

Trigger the SMR with the Beat Sequencer and use a high Q value to create a rhythmic "ping".

hear a filtered version of this noise. Adjust the sliders, use the **Rotate** encoder knob and **Spread** knob to bring in different bands of noise. Notice that, as you increase the Q value by turning the **Q (Res)** knob, the output slowly changes from filtered noise towards pure sine waves. This is the effect of a tight bandwidth (Q) or resonance where only very select frequencies from the noise source are allowed to pass through to the output. If you have a favorite noise module (or any complex sound source), try running it into the input jack(s).

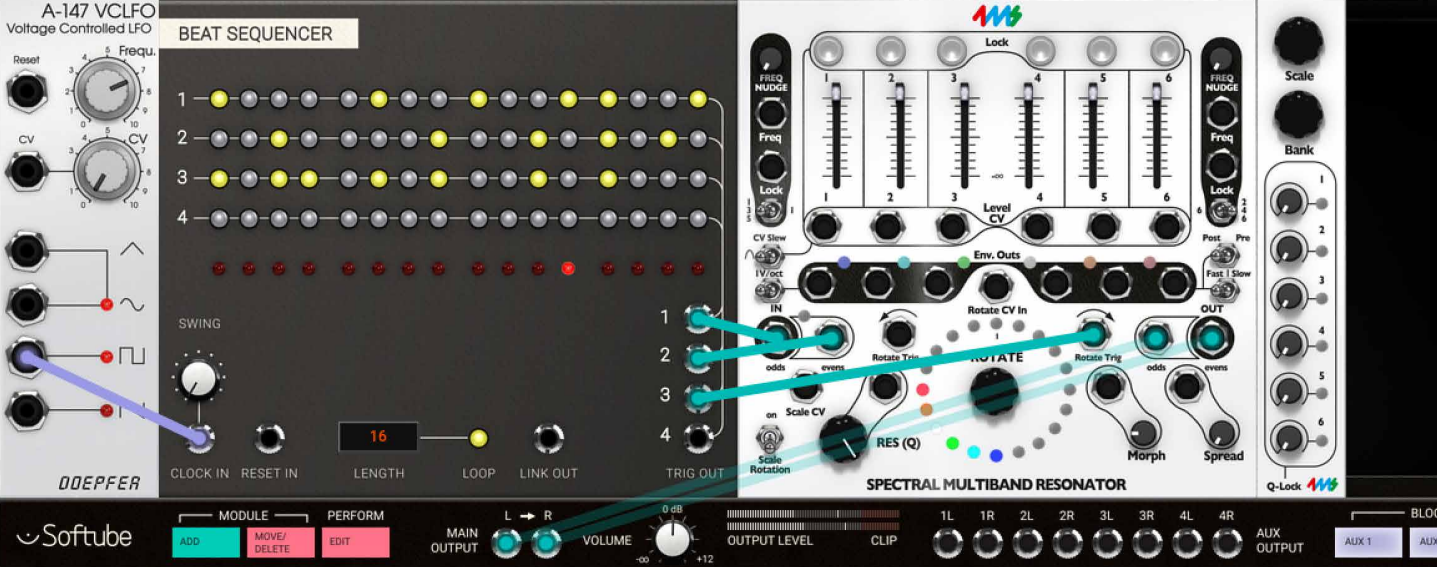
11. Finally, try running a 1V/oct melody line from a sequencer into one or both **Freq** jacks. Flip the **135 | 1** and **246 | 6** switches to select which channels are tracked, and which stay steady.

Tutorial 2: "Pinging" the Filter to Create Percussive Sounds

Preset patch "**Exp SMR tutorial 2**"

This is a great example on how to create resonant percussive sounds. You need a trigger or gate source from a MIDI TO CV or SEQUENCER module.

1. Patch the **Evens** output jack to main output left or, for stereo output, also patch the white **Odds** output jack to the main output right.
2. Patch your trigger sources into the **Odds** and **Evens** in jacks. Two sources with different rhythmic content is ideal, however if you only have one trigger source, just patch into the **Odds** in jack. In the preset example we have used the BEAT SEQUENCER driven from a LFO.
3. To start, have the triggers firing about once or twice per second. The odds IN jack goes to channels 1, 3, and 5. The evens IN jack goes to channels 2, 4, and 6. So each trigger source can strike a chord of three notes.
4. Listen to how the channels resonate when struck with a trigger or gate, they should sound like a gong or marimba. Adjust the **Q (Res)** knob to change how "ringy" the sound is: low Q is like a wood block, high Q is like a large bell. The amplitude of the triggers also effects the sound, try attenuating or boosting the triggers before they reach the SMR. Notice that the SMR will be struck on both the rising and falling edge of a gate, so two sounds will be heard for every gate. Triggers will cause only one sound since the rising and falling edges are very close together. Adjust the **Rotate** encoder and **Spread** knob to change the pitches of the notes, and play with the sliders to adjust the level of each note.
5. Input another trigger source into the **Rotate Trig** clockwise or **Rotate Trig** clockwise jacks. If you don't have another trigger source, you could input



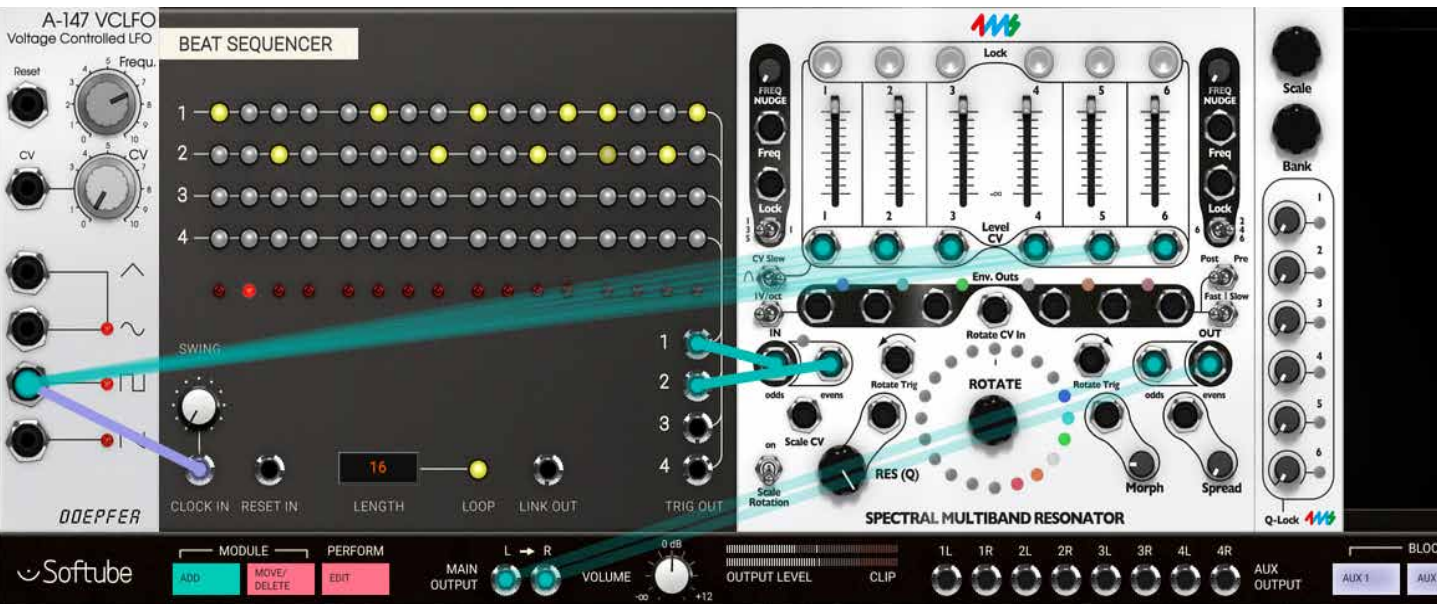
Play around with Rotate and Spread to change the pitches.

an LFO or envelope (0 to +5V) into **Rotate CV In** jack and the rotation will track the waveshape of the incoming CV.

Play with slowing down the rotation from the external triggers by increasing the **Morph** knob value. Try setting the rotation trigger source super fast and then turning **Morph** knob up and down to adjust the speed of the motion. **Morph** is a powerful way to limit musical movement (and it can be CV controlled).

6. Try out clicking on the **Scale** knob in the expander section, and hold down the mouse button. This will display the currently selected scale and bank by one or many blinking LEDs in the upper half, and a solidly lit lower half of the LED ring. Notice how the LEDs change: the bottom six LEDs are one color and there should be one LED that's slowly blinking colors in the top half of the ring. This is the currently selected scale. Click and turn the **Scale** encoder and the blinking LED will move: each position in the top half of the ring represents a different scale (there are eleven for every bank). Listen to how each scale

Use an LFO to set the channel volumes, adjust the CV Slew to get smooth out the CV waveforms.



Use an external sequencer to have a melody line track the pitches.

is different. If you like, patch CV into the **Scale CV** input jack to control the scale selection with another module.

7. Run an LFO square wave CV into the **Level CV** jacks. Now you can try flipping the **CV Slew** switch to the left to smooth out the click that occurs when a gate snaps up or down. Setting **Morph** higher when you flip the **Slew** switch will cause more slewing to happen, which means very fast CV in the **Level CV** jacks will be rolled off.

8. Try running a fast decaying envelope into one or both **Freq** jacks. Time it to the trigger sources to get membrane drum sounds. Or try a 1V/oct melody line from a sequencer to track the pitches. Try flipping the **135|1** and **246|6** switches to select which channels are modulated/tracked, and which stay steady.

Tutorial 3: Basic Remixing

Preset patch for Modular FX "**Exp SMR Tutorial 3**"

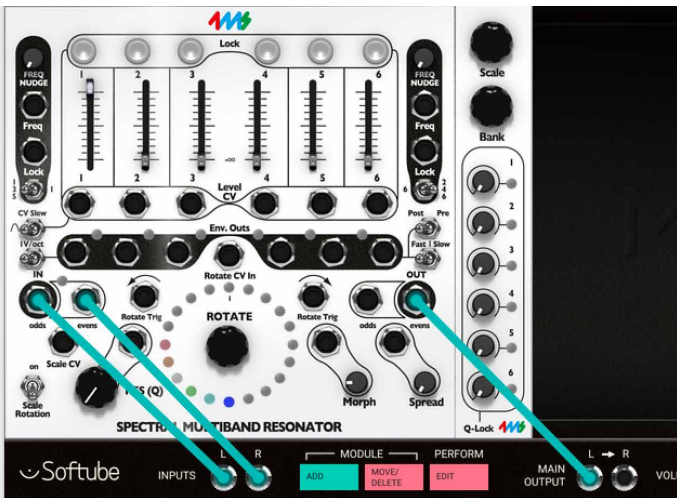
Remixing audio tracks often involves highlighting certain instrumentation and getting rid of others. This can be done by boosting and cutting frequency bands with the SMR and this how it is done:

1. Input a audio signal into the **In** jacks (use the **Odds** input jack for mono, or both jacks for stereo).

2. Take the output from the **Out** jacks to your mixer (use the **Evens** jack for mono, or both jacks for stereo).

3. Turn **Res (Q)** knob all the way down.

4. Click on the **Bank** encoder and the bottom six LEDs should change color, each color represents a different bank. Keep holding down the mouse button and keep changing by dragging until the you've selected the deep blue bank. Any bank will work, but the deep blue bank is nice for re-mixing because the scales are set to common graphic EQ frequencies.



Basic remixing. Connect the audio from the DAW to the SMR's inputs.

5. Slide channel one's **Volume** slider up and the rest of the sliders down. Turn the **Rotate** encoder to rotate channel 1 up and down the frequencies while you listen to the audio. Keep rotating until you find the frequency band you want to boost. Adjust the **Res (Q)** knob to narrow the band and add resonance if you wish. You may also want to adjust the left **Freq Nudge** knob to fine tune the frequency.

6. When you find the frequency setting you want, tap the channel's **Lock** button above the **Volume** slider. This will lock the rotation as well as the **Freq Nudge** setting. If you also want to lock the Resonance setting, adjust the top **Q-lock** knob on the expander section to the right hand side. The **Lock** button above the **Volume** slider will be lit if the channel is locked and on the expander section, the Q-lock LED will be lit if **Q-lock** is active (**Q-lock** knob is above zero). If you don't lock the Q, then the channel's resonance will be controlled by the **Res (Q)** knob and jack along with all the other channels with unlocked Q.

7. Repeat the process (steps 5-7) for channels 2 to 6. Note that for channels 2, 4, and 6 the **Freq Nudge** knob on the right is used, while for channels 1, 3, and 5, the **Freq Nudge** knob on the left is used. Make sure the **135 | 1** switch is set to 135, and the **246 | 6** switch is set to 246 (otherwise the respective **Freq Nudge** knobs will only control channels 1 and 6).

8. After setting all the channels to a useful frequency, adjust the mix of the six sliders. Play with adjusting the Q of each channel of the expander panel. You can range from essentially a re-EQ'ed track to harmonizing, resonant bowl sounds. Since each channel has a different Q value, both sounds can coexist.

9. To take the next step, consider running LFOs, envelopes, or clock gates into the **Level CV** jacks. Consider rotating some channels while keeping others still. Or run an LFO or envelope into the **Rotate CV** input. Play with **Morph** to make the rotation slower or faster. Another idea is to run a slow LFO into the **Freq** input, to create a filter sweep effect. Keep this patch going, and continue on to the next patch idea...

Tutorial 4: Advanced Re-Mix Techniques

Preset patch for Modular FX "**Exp SMR tutorial 4**"

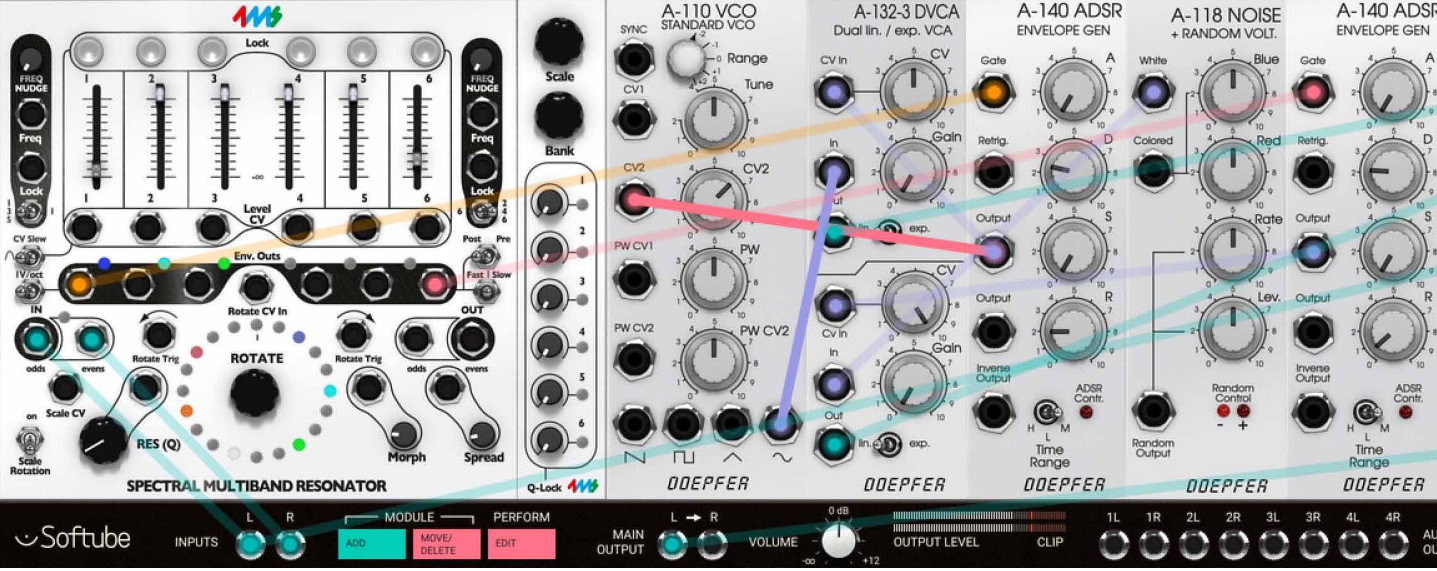
This patch idea is a powerful way to sync your modular system with an external audio source, whether it's a prerecorded track or a feed from other instruments being played live. The basic idea is to convert different frequency bands into trigger outputs, with the rhythm of the triggers matching the rhythm of the original audio's instrumentation.

1. Start with step 1 through 4 of the RE-MIX patch on the previous example: Patch audio inputs and outputs, turn down **Res Q** and select the EQ bank scale (deep blue).

2. Set the **Select** switch Odd (**135/1**) to 135, and the Select switch Even (**6/246**), to 246.

3. Set the Envelope follower out switch (**Fast/Trig/Slow**) to the center position (triggers), and set the Trigger detect switch (**Post/Pre**) to POST. The **Scale Rotation** switch should be turned off (down position).

4. Let's say you have a dance track with a kick, snare, hi-hat, and some melody lines happening. You can use the SMR to convert each kick drum hit into a trigger output, which you can use to trigger your own kick drum (or clock your master clock, or trigger anything on the modular). Start by patching the **Env Out** jack for channel 1 into something that makes sound when triggered.



Advanced remixing. Let the SMR trigger an external sound source, a Doepfer VCO in this case.

5. Turn down all the sliders and bring up the slider for channel 1 to around 50%. Listen to the SMR's output as well as the externally triggered module. You might want to monitor the input audio, too. Use a audio mixer and mix in your dry signal from MODULAR's inputs.

6. There are four parameters to play with to get beat-synced trigger outputs: **Rotation, Freq Nudge, Q,** and filter channel volume level. Turn **Rotate** so that channel 1 is triggering the external module whenever the kick drum hits. Probably for a typical kick drum, channel 1 should be on the 2nd, 3rd, or 4th note (it depends on the pitch of the kick you're trying to tune into). Set **Freq Nudge** to dial in a frequency in between two rotation spots. Turn **Q** up to narrow the band. Around 50% is a good place to start for syncing to drum sounds.

7. Set the **Level** slider to set the threshold at which the trigger fires. If you set it too high, the ENV OUT light will stay on, and it it's too low it'll stay off. If the slider seems to have no effect, double-check that the trigger detect switch (**Post/Pre**) is set to POST.

8. When you have found a good spot, use the lock function above the filter channel level slider to lock that frequency. Also experiment with different Q values for this particular channel on the expander module.

9. ADVANCED: Use an external CV source (try an LFO or a sequenced envelope) into the **Level CV** jack with the slider all the way up. This way you will have an automated mute/unmute of that filter channel.

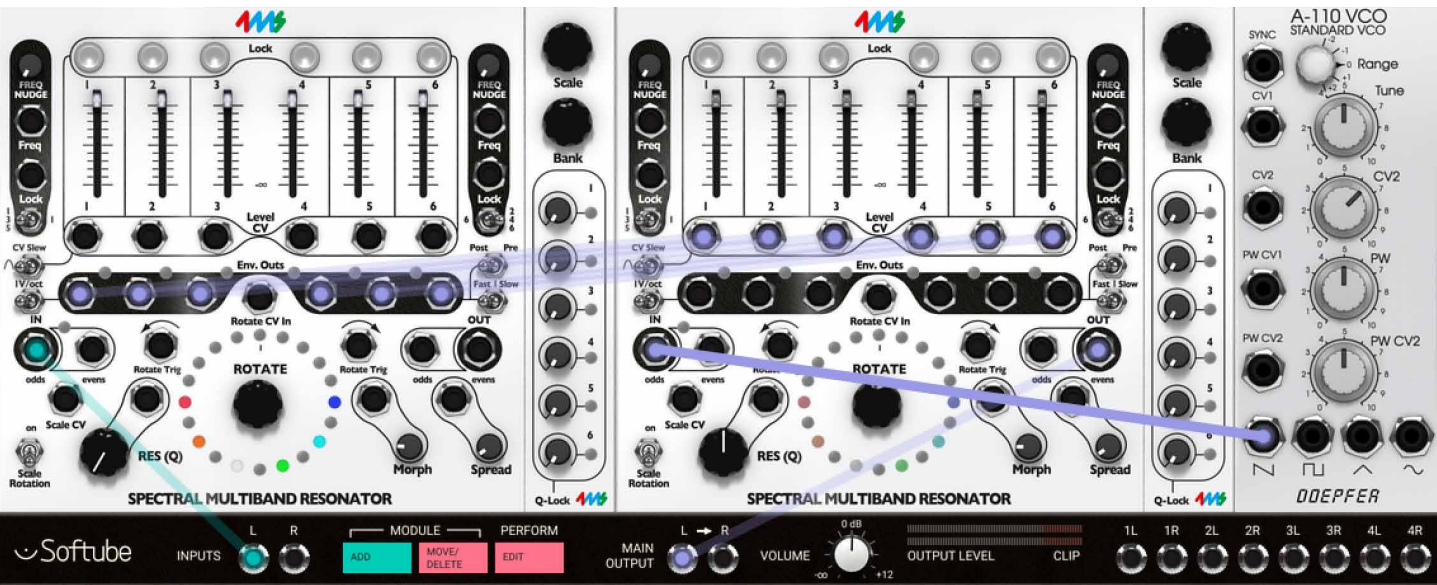
Tutorial 5: Vocoding and other spectral transfers

Preset patch for Modular FX "**Exp SMR Tutorial 5**"

Using two SMR's, you can create spectral transfer effects. One type of spectral transfer is known as *vocoding*, which is classically used with human speech and a sawtooth oscillator.

1. Patch a human speech audio signal into the first SMR's **Odds** input jack. Use a vocal sample, or a direct lined microphone from your audio interface. You are not limited to just using human speech, but it's a great way to start playing with this technique.
2. Patch the carrier signal into the second SMR's odds input. Use a sawtooth VCO, or perhaps an FM'ed VCO. You could also use another sound source or a complex sound by patching one signal in via the left jack and another via the right jack.
3. Patch the second SMR's **Evens Out** to the MODULAR **Main Out** so you can listen. You may also wish to patch the first SMR's **Evens Out** to a mixer and for monitoring the modulating signal.
4. Flip Trigger detect switch (**Post/Pre**) to POST on the first SMR.
5. Patch all six **Env Outs** jacks on the first SMR to the six **Level CV** input jacks on the second SMR. If this is your first time, keep them in the same order (1->1, 2->2...)
6. Play with **Rotation, Spread, Scale** and **Banks** of both SMRs. You might want to start in the blue bank (graphic EQ) and with the channels positioned as shown in the diagrams above.
7. Play with the **Res (Q)** knob of both SMRs. To fine tune the patch, follow the technique outlined in the RE-MIX patches, setting the note, scale, bank, and resonance of each channel on both SMRs.

Vocoding. Let one SMR track the sound sources, and another to re-synthesize it.



RES (Q)
(SCALE)
SPECTRAL MULTIBAND RESONATOR

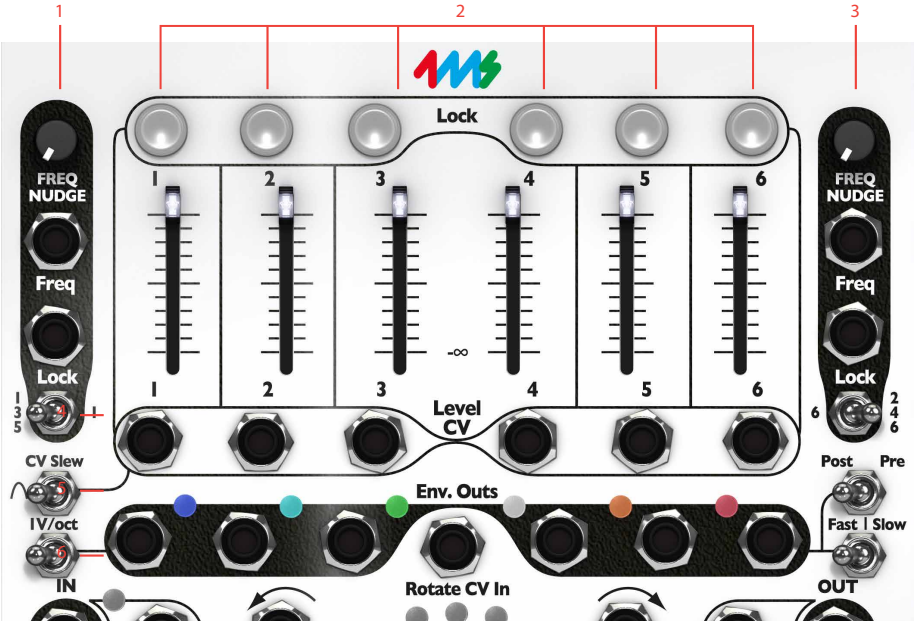
Parameters

Freq Nudge (odd bands) This knob detunes the odd bands: 1, 3 and 5, or just band 1, dependent on how the **135/1** switch is set. (See description below.) De-tunes up to 5 semitones in extreme clockwise position.

Lock Buttons 1 – 6 These toggle buttons are the frequency lock buttons for the different filter channels. When an unlit button is clicked once, it will lit up white (active). Once clicked again, it will turn unlit, grey, again (not active). When a channel is frequency locked it will not be affected by any change by the **Rotate** encoder knob, scale or bank encoder, nor will it be affected by any input frequency related CV (**Scale CV, Rotate CV, Spread, Nudge**) as long as it is activated. Notice that the **Lock** button state is the offset value for the **Lock Input CV** (see description below).

Freq Nudge (even channels) This knob de-tunes the even bands: 2, 4 and 6, or just band 6 , dependent on how the **6/246** switch is set. (See description below.) De-tunes up to 5 semi tones in extreme clockwise position.

Volume sliders 1- 6 These are the volume sliders for the different filter channels in the SMR, which control the volume out of that particular filter band for the odd and even outputs. These sliders will also affect the triggering levels for the envelope follower



1. Freq nudge (odd bands)
2. Lock buttons 1-6 and volume sliders 1-6
3. Freq nudge (even bands)
4. 135/1
5. CV Slew
6. 1V/oct
7. 6/246
8. Post/Pre
9. Fast/Trig/Slow

outputs when the **Trigger Detect Switch** is set to post (see further description below).

135/1 (Select Switch Odd) This switch determines whether or not all odd filter channels will be affected by the **Freq Nudge** (odd channels) and external CV through the **Freq (Odd)** input jack. When this switch is in "135" position, all odd channels are affected, but if switch is in "1" position only filter band 1 (blue), is affected.

6/246 (Select Switch Even) This switch determines whether or not all even filter channels will be affected by the **Freq Nudge** (even channels) and external CV through the **Freq (Even)** input jack. When this switch is in "6" position, only filter band 6 (red), is affected, but if switch is in "246" position, all even bands are affected.

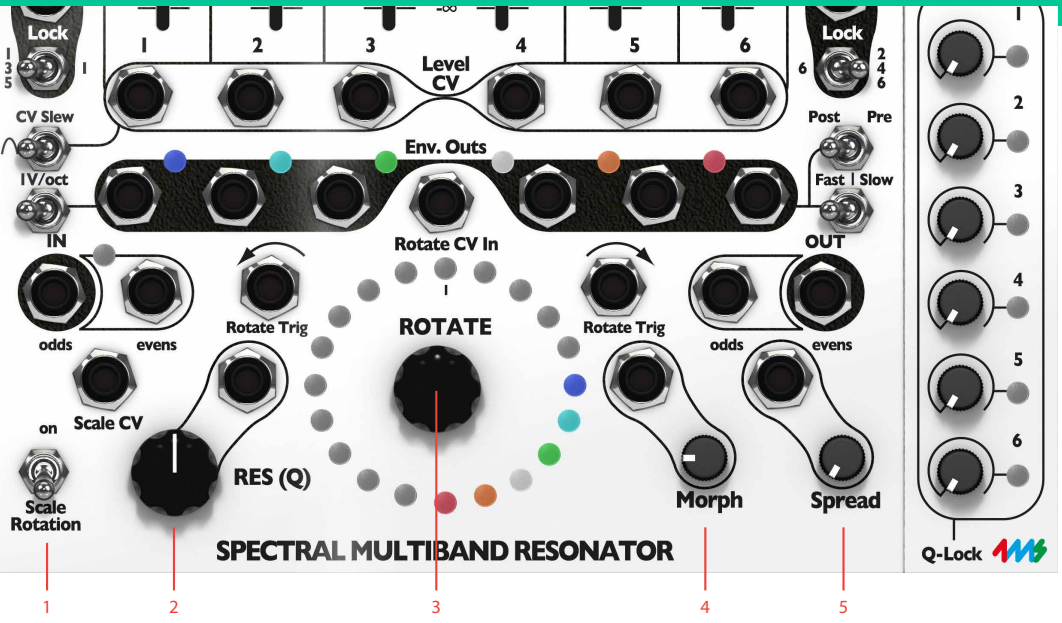
CV Slew The **CV Slew** switch can be set to soften the CV patched into the **Level CV 1-6** jacks. This is handy to avoid crackles and abrupt volume changes. **Slew** is engaged when the switch is in the left position. The slew rate is affected by the **Morph** knob position when the **Slew** switch is switched on in the left position.

Post/Pre (Trigger detect switch) The trigger detect switch, on the panel named **Post/Pre** sets wheth-

er envelope follower detection will occur before or after level set by the **Volume Sliders 1-6**. When set to "POST" position, the output of the envelope followers, **Env Outs 1-6**, will depend on the signal level set by the **Volume Slider 1-6** and **Level CV 1-6** inputs. When set to "PRE" position, envelope follower outputs will only reflect signal at the main audio inputs **In Odds** and **In Evens**.

1V/Oct (Env. Out) This switch changes the behavior of the envelope follower outputs, **Env Outs 1-6**; from envelope follower/trig functionality in "OFF" position, to quantized note CV reflecting the different channels in the "1 V/OCT" position. This functionality of the original hardware firmware version 5 is very handy when using the SMR as a quantizer or chords-generator.

Fast/Trig/Slow (Env. Follower Out) The envelope follower out switch, marked FAST/TRIG/SLOW on the SMR panel, sets the character and timing factor on the **Env Out 1-6** CV outputs. In FAST and SLOW mode the envelope follower will output a continuous CV reflecting the audio level of that particular band. FAST response time means that CV level will react quicker than in SLOW mode. When this switch is set to MID position, the **Env Out 1-6** will either output nothing or a high pulse when a



1. Scale rotation
2. Resonace
3. Rotate
4. Morph
5. Spread

certain level (-40dB) is exceeded and will remain so until signal has passed below this level again.

Scale Rotation switch When this switch is set to "ON" position, each revelation of the **Rotate** encoder, by the **Rotate Trig** (clockwise) or the **Rotate Trig** (counter-clockwise) inputs will continue on the next available scale within the bank, instead of starting over at the same scale. Each scale revelation starts at the 12 o'clock position.

Res (Q) This knob controls the overall resonance offset of the filters, from no resonance to a clear oscillation sine tone suitable for instrumentation. This knob setting will be added to the CV of the **Res Q** CV input. A filter band that has an active **Q-Lock** on will not be affected by this knob until its lock has been released.

Rotate The **Rotate** encoder lets you rotate the filter channels of the SMR around the presently selected 20 step scale. Filter bands rotated past 12 o'clock will be starting at the beginning of the scale again, or in case the **Scale Rotation** switch is set to "ON", when passing 12 o'clock it will go to the next scale in the bank. The response of the encoder is affected by the **Morph** knob (see description below).

Morph This knob sets the speed of the cross fade between the different steps in the selected rotated scale and affects all changes within that scale (the **Rotate** encoder, **Rotate CV**, **Rotate Trig Clockwise** and **Counter-clockwise**, **Spread**). This means that when **Morph** is set

fully counter-clockwise, changes to rotated scale will be very abrupt and instantaneous. When **Morph** knob is turned clockwise, the rate will change to slower and slower. When a change to the rotate encoder or any related rotate scale change is faster than the change rate of the **Morph**, there's a "memory" build into the **Morph** which will remember and play out the stored changes. This very apparent when for example sending triggers into the **Rotate Trig** jack at a pace that exceeds that of the **Morph** and then subsequently changing the **Morph** rate to a faster pace, the stored rotate steps will be rapidly played out. The stored **Morph** memory will however never be larger than 127 steps.

Spread The **Spread** knob set the spacing in steps between each filter band in the selected scale. When this knob is set fully counter-clockwise, the filter spacing will be 1 step, meaning that all bands will appear besides one another in the currently selected scale. When turned clockwise, the spacing between each step will increase successively by 1 step at a time, up until the channels are 14 steps apart. Evidently, this means that they already after 3 steps spacing will be spread beyond the 20 note scale and will "spill over" into the next scale upwards and downwards in the bank (if scale rotation switch is turned on). The **Spread** knob will work as an offset when simultaneously controlling **Spread** via the **Spread CV** input jack.

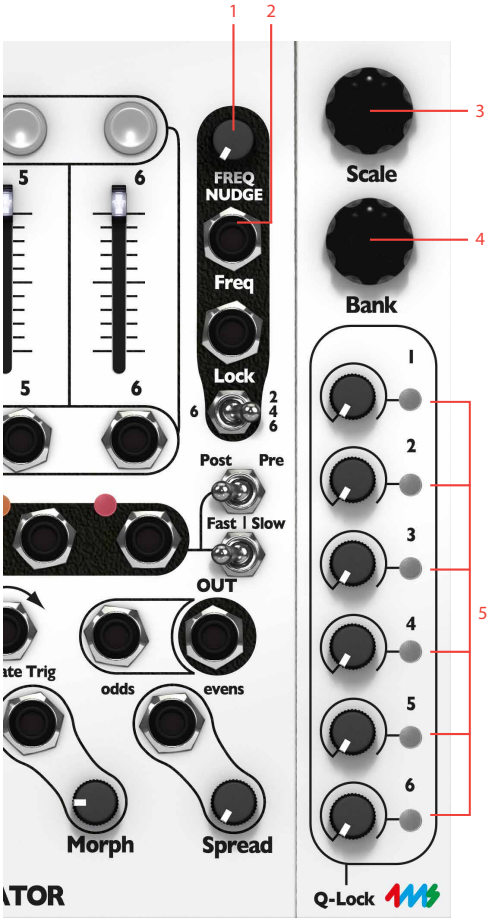
Expander Section

The expander section adds easier control over parameters that in the hardware are somewhat hidden.

Scale This encoder shifts all the filter channels (not presently locked and down the 11 available scales within the presently selected bank). The selected scale will be seen on the upper part of the encoder ring when clicking on and dragging the mouse cursor up and down. The colors on each of the different scale LEDs represents the channels assigned to that particular scale. If more than one band uses the same scale, that scale led will toggle between the different colors of the channels.

Bank This encoder shifts all bands available (not locked) between the different scale banks. More information on which bank colors represent which scale banks below.

Q-locks 1 – 6 The **Q-locks** are individual control over each filter channel's Q value. When a **Q-lock** is raised above 0% that band will not be affected by the **Res (Q)** knob or the **Res (Q) CV** input as long as it is locked to the value set on this expander section.



1. Lock button LEDs
2. Volume slider LEDs
3. Scale
4. Bank
5. Q-Locks 1-6 and Q-Lock LEDs

Indicators

Lock buttons LEDs Shows if a band is locked or not. When a band is locked it will not be affected by scale rotation movement of any kind (encoder or CV induced).

Volume Slider LEDs The intensity of these white LEDs shows the volume of the audio of the particular filter channel. A channel clipping will indicate this by flashing white light.

Env Out LEDs These LEDs reflect the output CV of the envelope follower on the **Env Out 1-6** jacks. They will pulsate when the envelope follower is in fast, slow or trigger mode. Note that they will still reflect the envelope follower output CV even though the 1V/OCT function is active.

Input Clip LED This indicator is lit when there's clipping on the SMR input.

The LED Ring This is the most important indicator on the SMR, which normally shows the 20 scale steps in the currently selected scale. The different colors each represent the filter channels 1-6. When the scale or bank encoder is activated by the user, this LED ring will also briefly show the currently selected scale and bank (until the user releases the mouse button).

Expander Section

Q-locks 1 – 6 LEDs The Q-locks LEDs indicate that a **Q-lock** for that particular channel is active, thus that particular channel is not affected by the main **Res (Q)** knob or **Res (Q) CV** input jack.

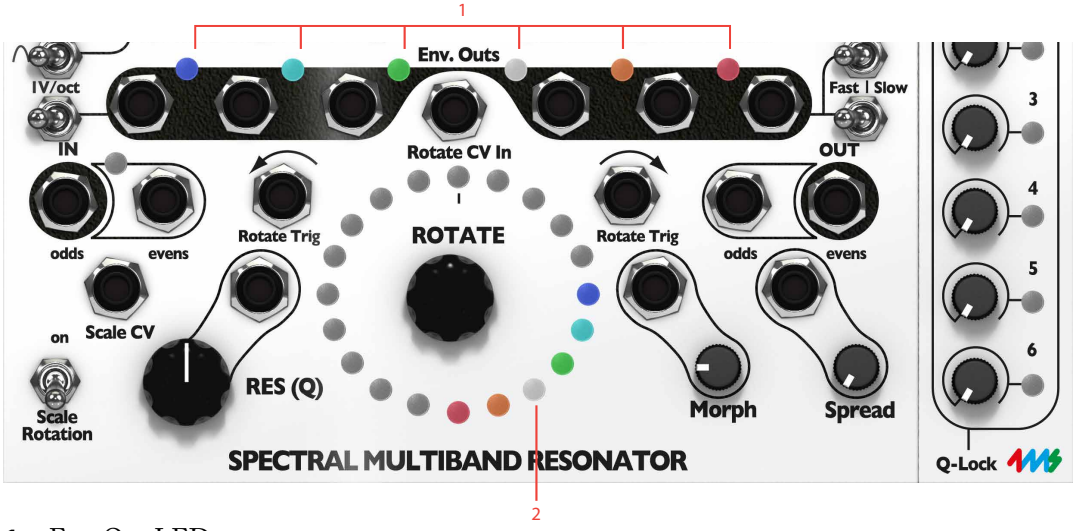
Inputs

Freq (Odd) This is the main 1 V/octave CV input for the ODD filter channels of the SMR. When the **135/1** switch (Select switch ODD) is set to "135" all odd channels will be affected by the CV input, but when the switch is set to "1", only the first band will be affected.

Freq (Even) This is the main 1 V/octave CV input for the EVEN filter channels of the SMR. When the **6/246** switch (Select switch EVEN) is set to "246" all even channels will be affected by the CV input, but when the switch is set to "6", only the last band, number six, will be affected.

Lock (Odd) This is the lock toggle input for the odd channels. It will trigger and toggle the lock function when a CV below 2.5V goes above that threshold. This function will affect the band currently selected via the **135/1** switch (select switch ODD). If a channel already has its lock button active, a pulse at this input will toggle it off.

Lock (Even) This is the lock toggle input for the even channels. It will trigger and toggle the lock function on a rising edge trigger (CV below 2.5V goes above that threshold). This function will affect the band currently se-



1. Env. Out LEDs
2. The LED ring

lected via the **6/246** switch (select switch EVEN). If a channel already has its lock button active, a pulse at this input will toggle it off.

Level CV 1- 6 These inputs are the CV inputs for controlling each band's level. When a CV signal is input here, the **Volume Sliders 1-6** will act as attenuators of the incoming CV, meaning that you will still be able to control the maximum volume of each band from the panel.

In Odds This is the common input for all odd filter channels. It is pre-patched to the **In Evens** jack when that jack is not in use. This means that the input signal goes through all filters, odds and evens, if no signal is patched into the **In Evens** jack.

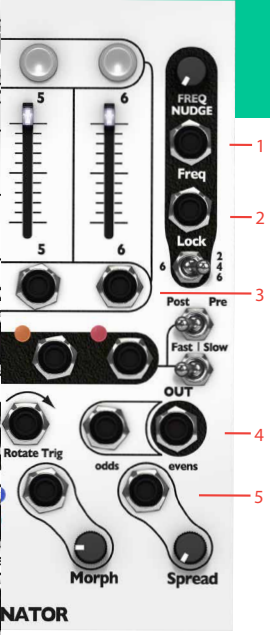
In Evens This is the common input for all even filter channels. When a signal is inserted here it will go through all even channels (2, 4 and 6). When no cable is inserted here it is pre-patched from the **In Odds** jack (see above).

Rotate Trig (Counter-clockwise) When a rising edge trigger (CV below 2.5V goes above that threshold) appears on this input, all channels not locked will shift one step back counter-clockwise. This is equivalent to turning the **Rotate** encoder one click to the left.

1. Freq (odd)
2. Lock (odd)
3. Level CV 1-6
4. In odds, In evens
5. Scale CV, Res (Q) CV

Rotate Trig (Clockwise) When a rising edge trigger (CV below 2.5V goes above that threshold) appears on this input, all channels not locked will shift one step forward clockwise. This is equivalent to turning the **Rotate** encoder one click to the right.

Scale CV Input CV on this jack will select one of the 11 available scales within the currently selected bank. Click on the Scale knob in the expander section to have a visual check on the what scale that is currently selected.



1. Freq (even)
2. Lock (even)
3. Level CV 1-6
4. Rotate Trig
5. Morph CV, Spread CV

Res (Q) CV This is the jack for external CV control of the resonance of all filter channels not currently locked by the **Q-locks** on the expander section.

Morph CV This jack is the external CV control of the morph parameter that can be offset via the **Morph** knob on the panel.

Spread CV The **Spread CV** jack is the external CV input for the spread parameter that can be offset via the **Spread** knob on the panel.

Outputs

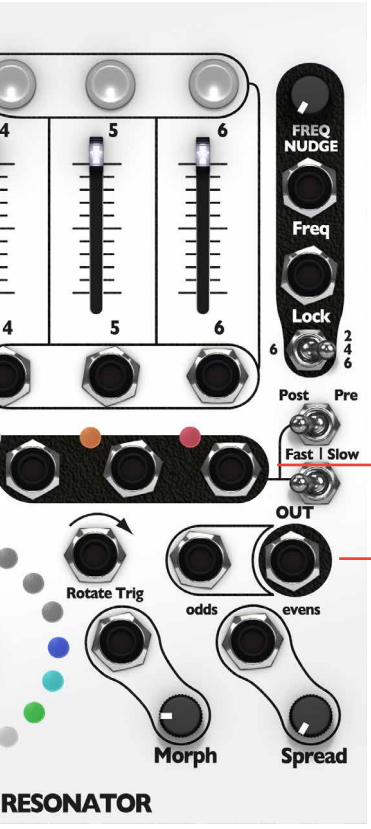
Env Outs 1 – 6 These outputs, **Env Out 1-6**, are the envelope follower outputs for each band. The outputs here are always control voltages (CV), not audio. When the **1V/oct** switch is set to "OFF", either envelope follower (FAST or SLOW) control voltage or triggers (TRIG mode) will appear on these jacks.

The trigger length on these outputs will depend on the set filter band level (**Level CV** input and **Volume Slider**) when the **Post/Pre** switch is set to "POST". When the **1V/oct** switch is set to "1V/OCT", each of these output jacks will output the quantized CV of each band's current cutoff frequency.

Out Odds This is the output jack of all odd filter channels. It is pre-patched to the **Out Evens** jack if no cable is patched into the jack (see description below).

Out Evens This is the main output jack of the SMR. If no cable is patched into the **Out Odds** jack, this is the output of all filter channels in the SMR.

Outputs



1. Env Outs 1-6
2. Out odds, Out evens

Scale Bank Colors

When turning the **Bank** encoder in the expander section, part of the LED ring will change color to indicate which scale bank that will be active when the mouse button is released.

1. Western scale (white)

Just intonation. Each scale contains a root note and one interval, repeated over 10 octaves. Each scale uses a different interval (m2 to M7) and spans 27.5Hz to 15kHz

2. Indian pentatonic (green)

Scales 1-8 have five notes per octave, spanning four octaves. Each is one octave above the previous. Scales 9-11 contain selected notes.

3. Alpha scale, selected notes (lavender).

Each scale contains intervals from Wendy Carlos's micro- tonal scale, spanning roughly from 20Hz to 20kHz

4. Alpha scale, selected notes (cyan).

More variations. All intervals are multiples of 78.0 cents (perfect fifth split into nine equal parts), so there are no octaves, yet rich harmonies.

5. Gamma scale, selected notes (orange)

Each scale contains intervals selected from Wendy Carlos's non- octave repeating Gamma scale, spanning roughly 20Hz to 20kHz.

6. 17-note per octave (yellow)

The first scale starts at 13.75Hz and each note is equally tempered to a 17-note octave. Each scale is one octave above the previous.

7. Chromatic (purple)

The first scale starts at E2 (82.4Hz) and each note is one semi-tone above the previous. Each scale spans an octave and a half and is six semi-tones above the previous.

8. Diatonic 1/2 (magenta)

Each scale is a diatonic scale spanning almost three octaves. The first two scales start on A, the next two start on A#, then B, C, C#, D. In each pair, the second scale is two octaves above the first.

9. Diatonic 2/2 (rose)

Continuation of previous bank, starting at D#, E, F, F#, G, G#

10. Western dual-interval (light green).

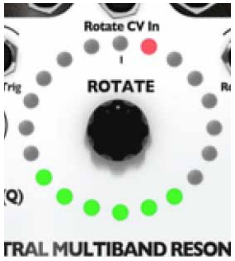
Each scale starts on G1 (49Hz), followed by two intervals. The triad is repeated across seven octaves, an octave higher each time. The pair of intervals varies per scale: M2/P5, M3/flat5, m3/M5, M4/P5, m3/#5, P4/P5, M3/M6, m3/flat6, M3/#5, m3/m7, P5/M7

11. Mesopotamian (Red)

Various ancient scales from Mesopotamia. Scales 1, 3, 5, 7, 9, and 11 have six notes starting on A2 (55Hz), and then repeats over 3 octaves. The remaining scales are copies of the previous scale, but three octaves higher.

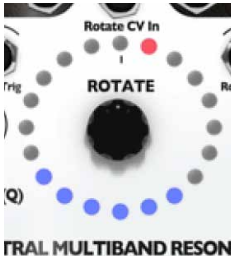
12. Shrutis, Indian scales (Yellow-green).

Uses 20 notes from a 22-note octave, so each scale spans one octave and is one octave higher than the previous scale.



13. Graphic EQ (blue)

These are common frequencies in Eqs. The first scale is 20Hz, 40, 60, 80, 100, 150, 250, 350, 500, 630, 800, 1k, 1.3k, 1.6k, 2k, 2.6k,3.5k,5k,8k,10k. Each scale is the same as the previous scale, shifted up about 3%

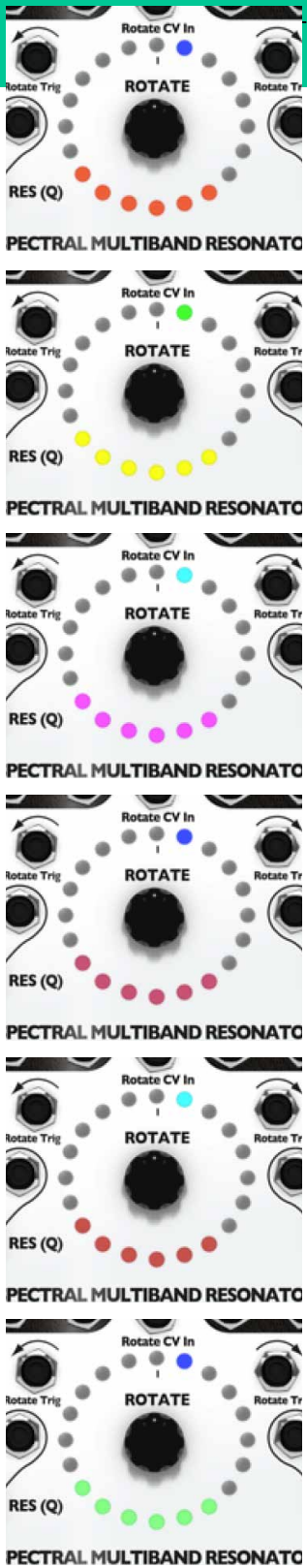


14. Gamelan (azure)

Scales 1 and 2 are the Gamelan Pelog (5-note) scale repeated over 4 octaves. Scales 3, 4, and 5 are Slendro (7-note) scales spanning 3 octaves each. Scales 6-11 are a selection of five notes from the Slendro scale.

15. Bohlen-Pierce (Lime)

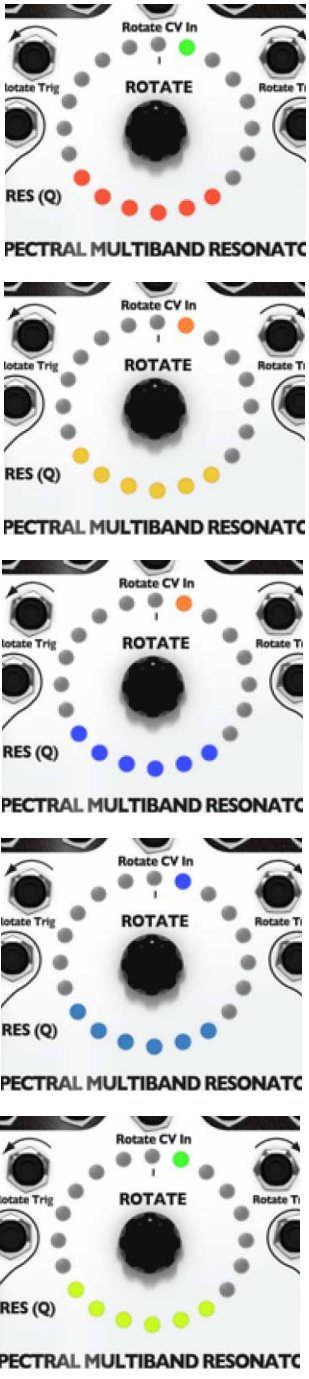
Instead of octaves, the BP scale is based on the tritave (triple the root frequency), with thirteen notes per tritave. Each scale starts on C1 and contains four selected notes per tritave, spanning five tritaves



In Use

As described in the tutorial examples and shown in the preset patches, the SMR is a very flexible module that can be used for a diversity of different tasks. For example:

- as oscillator bank for additive synthesis (marimbas, gongs, woodblocks)
- pulse triggering odd and even for different rhythms
- as vocoder
- as phaser
- as CV source/arpeggiator/sequencer
- as stereo-width thingy
- as multiband EQ
- as remix tool for extracting triggers from recorded material
- as quantizer
- as crazy ass effect
- as basis for multiband compressor patches
- as synth filter



Tip

Manually max the **Spread** knob and bring it down to ZERO in order to get your channels in order again, in case were they might have been set in different order via previous locks. The same technique can be used to bring all channels to the same scale by maxing and then setting the **Scale** knob to zero.

5 Abbey Road Studios Brilliance Pack

Foreword by Peter Cobbin

"Welcome to a little Abbey Road brilliance.

Throughout the '60s there were a number of small boxes dotted around Abbey Road Studios, which were simply known as 'Brilliance' or 'Presence' boxes. In this era, engineers were looking for ways of adding presence to their recordings, especially in popular music. Enter the Brilliance boxes: these simple passive equalizers were portable versions of the grey RS127s that were rack-mounted into the studio control room patchbays. They were designed to give Abbey Road Studios recording engineers additional frequencies that were not found on the EMI REDD studio mixing desks of the time.

Meet Lester

Lester Smith is a technical engineer at Abbey Road Studios and, amongst other things, is the custodian of our vintage equipment and microphone collection.

In recent years, while working on various Beatles and John Lennon-related mix projects, Lester introduced me to these little boxes. My assistants Mirek Stiles and Sam O'Kell had observed that, in various '60s setup sheets, the prevailing EQ was marked as "RS127". This equalizer was the 127th item made in-house by EMI for the Recording Sector. Keen to try these, Lester blew 40 years of dust from some of the boxes and wired them into our patchbay via the old 'Siemens' connectors and presto – instant zing! The large cut and boost control made it very easy to hear an immediate presence. Put simply – they're brilliant!

The Sixties were an adventurous period in our history. Rule books were put aside and considerable experimentation by engineers helped shape ways and means of producing sounds not heard before. Following this spirit of experimentation, our latest plug-in suite provides today's engineer and musician with a bundle of little boxes straight from this era.

The RS127s

The green and grey RS127s have identical circuits and controls. The grey has a chrome rack handle that made it convenient to plug into the control room patch-bays. There were usually two of these in each room, but due to their popularity additional EQ was often needed, and so stand-alone boxes were made to achieve this. These were painted green. Because of their portability, they were often used throughout the studio complex; not just in the studios themselves but also in the mastering, transfer and post-production rooms.

The RS127 is well documented in the highly recommended "Recording The Beatles" book (Kevin Ryan & Brian Kehew, Curvebender Publishing, 2006) which also refers to the change in line level and EMI standard impedance that has taken place since the early Sixties.

By happy accident, when comparing the green and grey units, we heard a dramatic difference when one of them was passed through an EMI interfacing transformer. The transformer "effect" exaggerated the EQ curves.

And so, in the spirit of the '60s, we have included this "not so precise" effect by providing two RS127 plug-ins: the grey rack version and the green standalone unit with the transformer effect. I have seen old setups where engineers used two RS127s in series for heavy EQ treatment, and I can highly recommend using these EQs on guitars, keyboards and vocals. If you find that +/-10 dB is not enough, do what they did back then – use two of them!

The RS135

The RS127 was primarily made to supplement the 5 kHz REDD desk EQ, and proved to be immensely useful and popular. However, there was still a need to boost frequencies between 5 kHz and 10 kHz.

EMI 8 kHz boost boxes were widely used but little was know about them. Only recently, when putting this project together, Lester removed the 8 kHz label (incidentally, made with a Dymo prototype) and found on the reverse that these boxes were in fact originally RS135s. Our researchers have shown that these boxes were designed for a 16.4 kHz boost, but modifications made at the time choked this frequency by half – enabling engineers to boost signals at 8 kHz. We have decided to add this to our 'Brilliance' suite, thus completing the range of presence EQs that were significantly used at Abbey Road Studios in the '60s.

Peter Cobbin, Director of Engineering, Abbey Road Studios April 2008

Introduction

The Brilliance Pack brings these classic EQs to modern audio workstations, providing authentic emulations of the hardware units.

The pack consists of three plug-in modules:

RS127 Rack

The RS127 Rack plug-in is a recreation of the original Brilliance Control rack modules which were installed in the studio control rooms.

RS127 Box

The RS127 Box plug-in recreates the standalone Brilliance Control and includes the exaggerated EQ curves caused when it is used with an EMI interfacing transformer.

RS135

The RS135 plug-in recreates the 8 kHz boost boxes, which were originally designed as 16.4 kHz boost units but later modified by Abbey Road engineers to work at 8 kHz.

RS127 Rack

The RS127 Rack plug-in recreates the original Brilliance Control. It boosts with a broad or 'blunt' bell-curve response and gives a 'medium blunt' bell-curve on the cut settings.

Frequency (kc/s) This control sets the centre frequency at which the plug-in will boost or cut in kilocycles per second - (kHz). The available settings are 2.7, 3.5 and 10 kc/s (kHz). The default setting is 3.5.

Gain (dBs) This control sets the amount of boost or cut which takes placed at the selected frequency in decibels. It can be set to +/- 10 dB in 2 dB steps. The default setting is 0.

RS127 Box

The 'RS127 Box' plug-in recreates the standalone RS127 unit along with effects caused by interaction between components in the RS127 and an interfacing transformer.

Frequency (kc/S) As on the RS127 Rack plug-in, this sets the centre frequency at which the plug-in will boost or cut in kilocycles per second (kHz). As on the RS127 Rack, the available settings are 2.7, 3.5 and 10 kc/s (kHz), but the interfacing transformer causes the centre frequencies to change and this effect has been replicated in the RS127 Box plug-in. The actual centre frequencies are 2.9 kHz, 4.2 kHz and 11.5 kHz.

Gain (dB) This control sets the amount of boost or cut which takes placed at the selected frequency in decibels. It can be set to +/- 10 dB in 2 dB steps. The default setting is 0.

The 'transformer effects' can be heard clearly at the higher boost settings, at which the plug- in gives a sharper 'Q' or bandwidth. At these settings, there is also significantly more boost than given by the same settings on the RS127 Rack plug-in.

RS135

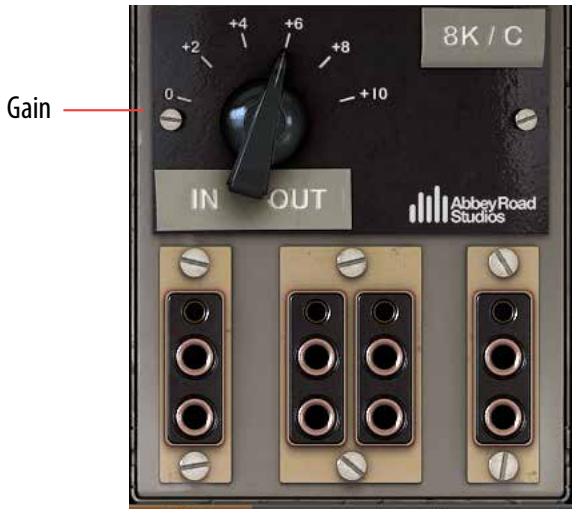
The RS135 plug-in provides an accurate model of the EMI 8 kHz boxes. It has a single control, which sets the amount of boost, and provides the same 'medium sharp' bell-shape characteristic as the original boxes.

Gain The Gain control sets the amount of boost in decibels. It provides up to 10 dB of boost in 2 dB steps. The default setting is 0.

Credits

Abbey Road Studios – product development and documentation, **Niklas Odelholm** – modeling, **Oscar Öberg** – DSP programming. **Torsten Gatu** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering.

EMI AND ABBEY ROAD ARE TRADEMARKS OF EMI (IP) LIMITED. REDD, RS127 AND RS135 ARE TRADEMARKS OF EMI (IP) LIMITED.



6 Acoustic Feedback and Acoustic Feedback Deluxe

Introduction

Real acoustic feedback appears when the sound waves from the cabinet hit the strings with enough energy (accomplished by playing loud enough) and with the right phase (accomplished by positioning the guitar at the correct distance from the cabinet). The difficult part with acoustic feedback on moderate volumes is therefore to get feedback on the desired notes—it is easy to get feedback, but if you for example need to get it on the high D, you have to find the exact distance so that the string vibrations and sound waves don't cancel each other. The rumour is that when Robert Fripp recorded the feedback parts on Bowie's *Heroes*, he made markings all over the floor in order to find the right position for different notes...

The Acoustic Feedback simulator is a mathematical model of real acoustic feedback. We have modeled how the sound waves hit the strings, how the strings interact with the amplifier and how the sound waves travel through the air back to the guitar again. But the only thing we couldn't model is the distance between the guitar player and the cabinet (computer?). So to solve this we made an "auto-positioner", pretty much like a Brian Eno running around with the cabinet so that Robert Fripp didn't have to move between the different markings on the floor.

So with all these parts put together, we present to you the Acoustic Feedback plug-in!

Getting Started

1. Make sure that you have a good guitar signal in to your audio interface. Use a line box or DI with a high impedance input (more than 500 kilohms).
2. If the guitar signal is too low (with peaks lower than 12 dB), use a volume plug in or compressor to boost the signal.
3. Insert the Acoustic Feedback on your guitar track in your favourite host software, and set all controls to the 12 o'clock position.
4. Insert Vintage Amp Room or your favorite amp modeler after the Acoustic Feedback.
5. Try some single string notes, such as the E on the 9th fret on the G string. Strike the note and wait for the feedback to appear. If it takes too long time, increase the **Feedback** knob. If the feedback is too fast and aggressive—decrease it instead. Bend the note to see how that affects the feedback.

Playing Techniques

The basic techniques in order to get the most natural sound from the Acoustic Feedback simulator are:

- *Don't push it*. If there isn't any energy left in your strings (ie. the strings have nearly stopped moving) the plugin won't be able to feedback on the correct note. End your notes after a while by muting the strings or pick another note. Don't let them ring infinitely.
- *No hum or noise*. Noise interferes with the feedback and shortens the lifetime of it. A short sustain (ie. dead strings) will also shorten the feedback lifetime.
- *Play nice*, single string and always end your notes deliberately. Sloppy playing is not rewarded!
- Notes played on low-numbered frets usually have longer sustain than notes that are played high up (for example above 15th fret).
- *No chords!*

If you do have a sloppy playing style, you will have to increase the **Tolerance** control to make the effect trigger more easily. The downside is that it doesn't always end very naturally.

> The feedback effect volume is independent of the input guitar volume! If you have a low guitar input volume, you will need to lower the Mix control in order to get a good balance between the clean guitar signal and the feedback effect.

1. Mix
2. Feedback
3. Tolerance and Tolerance indicator

User Interface

Mix	The mix between the DRY (original) and WET (feedback) guitar sound.
Feedback	Adjusts how aggressive the feedback is
Tolerance	Adjusts how easy it is to achieve feedback (and how tolerant it is with sloppy playing.
Tolerance Indicator	When the light is on, you got full feedback. When it starts to fade it's time to end the note.

Mix

Controls the balance between the feedback and the original signal. Outputs only the clean guitar sound when set on DRY, and only the feedback effect when set to WET.

Set this control so that the balance between the guitar and the feedback sounds natural. This knob is usually set somewhere in the green area.

> Try playing around with **Mix** set fully on WET. You can get some pretty interesting results from that...

Feedback

Controls how fast the feedback builds up. This corresponds in the real world to the volume of the real amplifier. Low settings will give subtle harmonics, and high settings will give an aggressive feedback.

Feedback = SUBTLE (Yellow area): The feedback will usually not buildup and the effect will be very subtle.

Feedback = NATURAL (Green area): The feedback will have a slow buildup time and act quite naturally, although it might be hard to get feedback on all notes.

Feedback = WILD (Red area): The feedback will have a faster buildup time and it will be quite easy to get the feedback effect on most notes, but sometimes it won't sound very natural.

> How the **Feedback** control works depends greatly on your guitar, preamp, volume, what note you play etc.

Tolerance

Controls how easy it is to maintain the feedback while changing notes, bending, doing vibratos etc. This corresponds in the real world to a combination of the distance between the speaker and guitar, but also the volume of the amp. When set at the lowest setting, it will only feedback on stable notes, and when set at the highest setting, it will feedback on most things, even noise or background hum.

Tolerance = MODERATE (Yellow): Feedback only on very stable notes.

Tolerance = NORMAL (Green): Feedback on vibratos, bends and slides. This usually gives the most natural sounding feedback.

Tolerance = RAMPANT (Red): Wild and unpredictable feedback. Sometimes.

If you find it too hard to get feedback, you can do one of three things:

1. Increase the **Feedback** control (to make the effect appear faster)
2. Increase the **Tolerance** control (to make the effect more tolerant towards note changes and different playing techniques.
3. Increase the input volume by increasing the gain of your preamp or using a volume plugin before Acoustic Feedback.

> The timbre of the feedbacking tone will change slightly if you change the **Tolerance** control during a feedback. This can sound abrupt and unnatural, so it's better to only change the **Tolerance** when no note is feedbacking.

Tolerance Indicator

The **Tolerance Indicator** indicates when a feedback is on its way, and when the feedback is about to die. If the **Tolerance Indicator** starts to fade, make sure that you end your note deliberately, before any strange sounds appear. This is specially helpful when you have a high **Feedback** or **Tolerance** setting.

Harmonic Selector (Deluxe only)

Controls which harmonic that will dominate the feedback. In the real world, this can sometimes be controlled by changing the distance from your guitar to the amplifier. In a world built up by zeros and ones, this is done by pushing a button (or five).

16' = SUBOCTAVE. This knob will add a subtle harmonic one octave below the fundamental. If only this harmonic is selected, the plugin will never start a feedback, but if it's selected in conjunction with other harmonics you'll get a very growly feedback. Use this as a you would use the "nitro" button in your favourite car racing video game.

8' = FUNDAMENTAL. This will give you feedback on the fundamental, ie., the same note as you are playing.

5 1/3' = ONE FIFTH ABOVE FUNDAMENTAL. This will give you a feedback on the fifth above the fundamental. Works best in conjunction with the 16' button.

4' = ONE OCTAVE ABOVE fundamental

2 2/3' = ONE OCTAVE AND A FIFTH ABOVE fundamental.

Using MIDI

Although the Acoustic Feedback doesn't have any built-in MIDI support, it supports automation, and is therefore easy to control via MIDI in a number of hosts. The most effective (and fun) way to control the Acoustic Feedback is by using an expression control pedal.

Expression Pedal

It is very easy to control the feedback effect by assigning the **Feedback** control to an expression pedal (a MIDI volume pedal).

Connections

If you don't have an expression pedal connected to your system, you will need:

1. Expression pedal. The cheap ones cost about €30 and work perfectly fine.
2. Expression pedal input. These connectors are usually found on MIDI keyboards, so take a look at the back of your keyboard to see if you have a 1/4" stereo jack with "Expression" or "Foot Control" written on it.

Connect the expression pedal to the MIDI device, and use the MIDI control section in your host software to assign the control number and MIDI channel to the feedback effect.

Feedback control

1. Assign the expression pedal to the **Feedback** control.
2. Set the **Tolerance** control to 1 (NATURAL).
3. When ever you want to get the feedback started, push the pedal (hence increasing the **Feedback** control).
4. The feedback effect will decrease as soon as you lower the **Feedback** control.

MIDI Automation and Touch-Ups

To really get the sound you want without controlling it with an expression pedal, you might want to use the automation feature in your audio sequencer.

1. Start out by setting the Acoustic Feedback on a setting that you find works most of the time and record your guitar track.
2. For parts that you don't want any feedback on, set the **Feedback** control to 0%.
3. On parts where you desperately need feedback, but your current setting doesn't give it to you, set the **Tolerance** to MAX, and gradually increase the **Feedback** during the duration of the note you're playing.
4. Just before the note ends, do a quick fade out with the **Feedback** control, and if necessary set the **Tolerance** control back to it's previous settings.

This way you can have "uncontrollable" settings, but by just using them on specific notes, you will avoid troublesome note endings and strange noises.

> Sometimes it is easier to use the **Mix** control to fade the effect in and out, but the **Feedback** control will always give more realistic results.

FAQ

I don't get any feedback? What am I doing wrong?

It could depend on a lot of things, but you can start by setting the all controls fully clockwise; **Mix** = WET, **Feedback** = WILD and **Tolerance** = RAMPANT. Then you will only hear the feedback effect with the most aggressive settings on the feedback algorithm. Make sure that you have enough input gain on your guitar signal, and strike a single note, for instance the open B string, and listen for the feedback.

When you have learned to get the feedback, decrease the controls one by one until you get a natural sounding feedback.

The feedback is too aggressive and doesn't build up in a realistic way

If the feedback comes too fast, decrease the **Feedback** control. It can be a bit difficult to find that sweetspot since it depends on your guitar, but it's usually located somewhere in the green area.

The feedback doesn't blend in with the sound

This has probably to do with one of two things:

1. The **Mix** control is set too high so the feedback effect is too loud compared to the guitar signal. Decrease the **Mix** control.
2. The **Feedback** control is set too high, so that the feedback doesn't build up naturally. Decrease the control.

The feedback doesn't stop when I change notes

If the **Tolerance** switch is set at RAMPANT, lower it to NATURAL or MODERATE. The **Tolerance** controls how easy the feedback aborts when you change notes.

If it doesn't help, and you really want the feedback to abort, make sure that you strike the strings when you change notes. Hammer-ons or pull-offs are not always enough to abort the feedback effect.

The feedback ends in mysterious ways...

Decrease the **Tolerance** control and make sure that you end your notes deliberately. If you mute the strings using your left hand, make sure that you don't end up with a harmonic instead. If you for example play on the 13th fret, you might end up with a 12th fret harmonic when you mute the strings, which will cause the plugin to feedback on the 12th fret harmonic note.

How to feedback on chords?

You don't. Well, sometimes it works, sometimes it doesn't. If you need a feedback on a chord, you could record it by using two takes. On the first take you play the chord without the feedback effect, and on the second take you play a single string from the chord with the feedback effect. With some creative mixing you can make this sound very realistic.

How do I control what harmonic the effect feedbacks on?

There aren't any "intelligent" algorithms to decide harmonic, it's more of a "survival-of-the-fittest" kind of thing. The strongest harmonic content of the note from your guitar usually survives in the end. The basic rule is that lower notes usually have a high harmonic feedback, and high notes have a low harmonic (or the fundamental) feedback.

How do I get that big Woodstock sound?

The Acoustic Feedback together with the White Amp from Vintage Amp Room is a good start, you can make the amp sound a little larger by setting the mic in a far-field position, but to really get that arena feeling you will have to add some stereo slap-back delays and a bit of stadium reverb.

Credits

Ulf Kilberg – mathematical modeling and research. **Niklas Odelholm** – modeling and sound design. **Oscar Öberg** – modeling. **Torsten Gatu** – interface and framework programming. **Arvid Rosén** – framework programming. **Per Connman** – prototype modeling. **Ulf Ekelöf** – 3D rendering and graphics.

7 Active Equalizer

THE ACTIVE EQUALIZER IS MODELED from a Swiss console equalizer, a Filtek Labo mk.5, with three adjustable bands, a low cut, and a high cut. It is a very solid construction and is very different from the Passive Equalizer. Where the knobs of the Passive Equalizer all influence each other (much like the tonestack of a guitar amp), the Filtek was built to minimize such effects. For instance, if you set the frequencies of the Low and Mid-filters so that they overlap, and cut both bands at –16dB, you won't get a doubled effect (–32dB) – you will still just have a cut at around 16dB. This is very different from how other (especially digital) equalizers usually work. And it is of course a big part of its sound.

For more info about this product, please see the chapter about the Focusing Equalizer.

User Interface

The Active Equalizer has three full parametric bands, a low frequency band (50-500 Hz), a mid band (380-3800 Hz) and a high frequency band (1.8-18 kHz) all controlled with a gain control (-16 to 16 dB), a frequency selector and a width/bypass switch. You also get two cut filters (18 dB/oct) and an output volume control.

1. Low band
2. Mid band
3. High band
4. Output
5. Gain control
6. Frequency select
7. Width/Bypass select
8. Low cut
9. High cut

Knobs

The three bands are equal, apart from their frequency ranges, and they all contain a gain control (labeled "+ dB –"), a frequency control (labeled "Hz" for the Low and Mid band, and "kHz" for the High band), and a width and bypass control (the switch beneath the knobs).

Gain Control Boost or cut in 2 dB steps (from –16 to +16 dB). Please note that all knobs are placed as if they are up side down. It isn't our fault – the original looked that way!

Frequency Select Use this to set the center frequency of the bands. Unlike most of our stuff, where we model a real unit with all its quirks and the labels never match reality, this sturdy unit actually does what it say it does!

Width/Bypass Select The switch lets you select between a sharp curve (leftmost position), BYPASS (middle position) and a blunt curve (rightmost position).

Low Cut An 18dB/octave filter with selectable bypass, ranging from BYPASS, 80 Hz, 180, 240, 320 to 500 Hz.

High Cut An 18dB/octave filter with selectable bypass, ranging from 5 kHz, 8, 10, 12, 15 kHz and finally BYPASS in the 3 o'clock position.

Output Output volume. Goes from nothing to +12 dB. At the 12 o'clock position, it's set at 0 dB, and going clock wise will increase the output volume by +2dB for each mark.

Please note that all gain controls range from –16 to +16dB, which is a quite big range, so please take it easy with the gain controls. In many cases, 6 or 9dB is the maximum gain you'll need.

Buying Recommendations

If you like the sound of the Active Equalizer and would like to get your hands on it, you should look for the Filtek Labo mk.5 console equalizer. It's a remarkably small and versatile equalizer with a very distinct sound.

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Credits

Oscar Öberg – modeling. **Arvid Rosén** – modeling. **Niklas Odelholm** – graphics design. **Torsten Gatu** – concept. **Ulf Ekelöf** – graphics rendering. Thanks to **Stefan Fandén** and the crew at Deluxe Music for letting us borrow the gear!

8 American Class A

Wide-open, airy, living electronics. If you will it, it is no dream with the American Class A. Based on the greatest classics of American mixing console hardware, this channel strip is fast and easy to work with, and brings a truckload of character to the party. With a shape section featuring gate and transient designing, a stellar four-band EQ with proportional Q and range control, a fascinating compressor design which can be switched between feedback and feed forward operations, and a warm, crunchy output drive section, no modern studio is complete until this sonic option is at hand.

Design Philosophy

The American Class A follows a format very familiar to Console 1 users, and this plug-in is indeed also a Console 1 channel strip. While we wanted to stay true to the original's sound and character, we also wanted to expand on some functionality, like the Shape and Drive sections, and get a perfect combination of versatility and character. You may think that a stepped EQ with seven frequencies per band is an unnecessary limitation in today's digital world, but a lot of the character of this console comes from the way you use it, and limited options in the EQ force you to bring out that character!

The compressor topology is also a major part of the sound. It's punchy and firm and has a unique style. It simply sounds fantastic! By making the shape section from the same topology and circuits as the compressor, we made sure that the tone of the original is always present, regardless of which part you use.

All in all, we're very proud of this. It features both extremely fine models of the hardware, as well as new concoctions, such as SHAPE and **Drive Character,** but they all breathe the same air and speak the same language. And on top of that, we added some nice visualizations. What's not to like?*

User Interface

For a Console 1 user, the user interface should feel very familiar. It consists of 5 sections; INPUT, SHAPE, EQUALIZER, COMPRESSOR, OUTPUT, and a large screen that visualizes the audio and settings.

On top of the screen you have the display section, which can be switched between WAVE and CURVES.

* Well, if you don't like the sound of the original console, we don't think you are going to like the sound of American Class A...

Input Meter Shows the peak level of the input signal (post **In Gain**). The top line is the peak hold level, white bar is peak level, and blue shows the RMS level.

Gate Curve Shows the knee of the **Gate**. The meter beneath displays the gain change of the entire Shape section. 0 dB is in the middle, and if you increase **Sustain** or **Punch** you'll see that the white meter is showing a positive gain change (moves to the right), which means that you're increasing the audio level. Turn on the **Gate** and you see a negative gain change.

EQ Curve Shows the EQ settings as well as a frequency analyzer view of the output signal. For stereo signals, the analyzer displays the left signal in white and right in red/orange ("red is right", as always!)

Compressor Curve Shows the knee for the compressor, as well as the current gain reduction. 0 dB (no gain reduction) is all the way to the right.

Output Meter Works just like the input meter, but on the output. Output meter sits after the Volume.

View Wave and View Curves By clicking on **View Curves** you'll get the above mentioned graphics, select **View Wave** and you'll get a wave form view. The wave form displays four different waveforms:

1. INPUT PEAK (grey)
   The Input Peak level shows your input level (post In Gain control). For stereo audio it shows the maximum of the two channels.
2. OUTPUT PEAK (blue)
   Same as Input Peak, but for the output.
3. RMS LEVEL (white line)
   The RMS of the output signal.
4. GAIN CHANGE (red)
   The combination of the gain reduction from the compressor and the gain change from the Shape section.

Input Section

Aside from the normal controls, **Low Cut, High Cut, Phase Invert**, and **In Gain**, you can also turn on **Filt To Comp** to filter the audio that goes to the compressor's sidechain.

Parameters

LP Low pass filter, 500 to 20000 Hz. Low pass filter is disabled in its max position.

HP High pass filter, 20 to 600 Hz. High pass filter is disabled in its minimum position.

Filt To Comp Engage to filter the compressor's side chain. This means that the filters will no longer affect the main signal path, and will only be used to send to the compressor. If you also need the filters in the main signal path you can use the filters in the EQUALIZER section.

Phase Invert Engage to invert the phase of the input.

Shape Section

The shape section is not modeled from existing hardware. We have instead chosen to use the compressor hardware to build our own gate and transient shaper. That gives us a shaper that's extremely versatile but has the same unique DNA and sound as the compressor. Technically speaking, we've used the very special RMS detector, VCA and circuit solutions from the compressor, and built a completely new thing out of it!

Parameters

Gate	Sets the gate's threshold. Disabled when set to its minimum position.
Gate Release	Sets the release time for the gate, from 0.3 to 3 seconds.
Hard Gate	Adjusts the ratio and knee of the gate. **Hard Gate** has higher ratio and sharper knee, and is more suitable for percussive sounds.
Sustain	Increase to add more sustain to the audio.
Punch	Strengthens or softens the sound's transient, the onset of the sound.
Ext S/C	When engaged, the external sidechain is routed to the GATE part of the SHAPE section.
Shape On	Engages or bypasses the SHAPE section.

Equalizer Section

The equalizer is based on a mastering grade 19" equalizer, instead of the console equalizer. With seven frequencies and 11 gain settings per band, the four bands offer great versatility, but at the same time force you to keep within the same limits as the original console, thereby giving you the same classic sound. For more fine tuning and mastering purposes, the original 19" unit has a **Gain Multiplier** which allows you to use finer steps than the 2-3 dB steps offered by the console version.

Parameters

Frequency	Sets the frequency of the current band
Gain	Sets the gain of the current band. The Gain has no effect if the LOW or HIGH BAND is set to "CUT".
Low/High Type	Select between shelving, bell, or cut filters.
Gain Multiplier	Sets the steps between different gain settings. At x1 the gain can be set to +/- 0, 2, 4, 6, 9, and 12 dB. At x0.5 it can be set to +/- 0, 1, 2, 3, 4.5, 6, and at x0.25 it can be set to +/- 0, 0.5, 1.0, 1.5, 2.25, and 3 dB.

Please note that the parameter value read-out always corresponds to the x1 setting.

Order Sets the order of the three main sections, SHAPE, EQ, and COMPRESSOR. SHAPE will always come before COMPRESSOR, but the position of the EQ can be shifted to before, in the middle of, and after the dynamics processors.

Equalizer On Engages or bypasses the EQUALIZER section.

Compressor Section

The compressor is faithfully modeled from the console compressor and has a couple of interesting features. The **New Type** switch allows you to switch between feed forward

("NEW") and feedback ("OLD") types of compression. In feed forward mode you get a very predictable and firm compression, while the feedback mode has a lot of overshoots and slightly more unpredictable results. The feedback mode sound is punchier and more aggressive, while feed forward is more suited to bus compression duties, vocals, etc.

Parameters

Release Sets the release time of the compressor, from 0.3 to 3 seconds.

Ratio Sets the ratio of the compressor. In general, the actual ratio is often less than what the panel says it is.

Parallel Engages the parallel compression.

Threshold Sets the threshold of the compressor, from +10 to -20 dB.

Hard Knee Adjusts the knee of the threshold. A hard knee has a sharper transition between compressing and non-compressing states. A soft knee as a gradual transition where the ratio gradually increases around the threshold. For most applications, **Soft Knee** is where you want to set it.

New Type Switches between feed forward ("NEW") and feedback ("OLD") modes. Feed forward is generally smoother and more predictable, feedback is punchier and more aggressive.

Ext SC Routes the external sidechain to the compressor.

Compressor On Engages or bypasses the COMPRESSOR section.

Output Section

The output section features the distortion from the preamplifers of the original unit, with an added **Drive Character** feature that allows you to control the tonality of the distortion. The original distortion can be a bit "fizzy", and by increasing **Drive Character** you'll get a more solid and fat tone.

Parameters

Drive Increase for more distortion. At "5" it will start to clip at 0 dB, pre output volume. Turn it off completely by setting it at 0.

Drive Character Increase for more high frequency distortion. At the minimum position (-10), you'll get a thick low frequency distortion, which has the same sort of character that a really underspecified, worn-out, transformer has. At +10 it's a "normal" full band distortion.

It is very effective to tweak the Drive Character according to the source material, if you for instance have a vocal track, you might want to set it so that the body of the vocals distort, but the "s"-sounds remain undistorted. You do that by starting from the minimum position and slowly turn up Drive Character. The sweet spot is usually somewhere between -3 and +3.

The Drive Character is not a part of the original hardware, and if you want it as "true to the original" as possible, you should set it at its maximum value (+10).

Volume Adjust the output volume of the plug-in.

Credits

Kim Larsson – modeling. **Björn Rödseth** – modeling and graphics programming. **Patrik Holmström** – framework programming. **Niklas Odelholm** – sound and graphics design, modeling, and project management. **Ulf Ekelöf** – 3D modeling. **Pavel Shyrinskykh** – presets.

9 Bass Amp Room

Introduction

WHEN WE DESIGNED BASS AMP ROOM we had two goals in mind. The first goal was to give the user the best possible sound quality and state-of-the-art physical modeling. The other, equally important goal was to give the user a plugin that was intuitive and allowed for a really fast work flow. Time is money, but cutting down on time shouldn't have to mean that you need to accept poor results. At least not with Bass Amp Room. And having an amp model that behaves just like the real thing will definitely speed up the work process, since you probably already know how to use it.

In short, you need the same skills to master this software as you need when dealing with the real studio setup. You can get back to doing what you do best, because music production should be about using your ears, not your computer skills.

WE HAVE IMAGINED TWO TYPES OF USERS. The first wants the best possible amp modeling and is willing to spend time tweaking the sound of each bass track. By working the mics, amp and D.I. tone controls, and, perhaps most importantly, the blend between the D.I. and amp, this user gets full control over the sound without ever losing it's authenticity.

The second user is one who, just like the previous user, wants the best possible amp modeling, but recognizes the fact that he/she cannot spend too much time tweaking it. That is why we have, for example, only a single tone control knob on the D.I. and a one-knob limiter. Don't be fooled by the simplicity, however; you still have the ability to create a unique signature sound. The strength lies in the very easy individual blocks (the amp, the D.I., the cabs and mics) and the ability to mix and match between them.

User Interface

Apart from a good sound, work flow is a crucial element of our design – the amps and mics should work just as they do in a real studio. We have tried to imi tate this as much as it is possible in a computer environment, and if you have experience working in real studios, you will notice that Bass Amp Room handles and sounds just the way you expect it to.

Amp Panel (Top Area)

In the Amp Panel, you can alter the settings of the amp by clicking the knob and dragging the mouse up and down, or left to right. Switches can be switched either by clicking on them, or by clickand dragging the mouse. In some hosts you can change the behavior of the mouse, but the default behavior of Bass Amp Room is the one described above.

Room View (Middle Area)

In the Room View you can do two things: select the other cabinet or change the mic's position.

Mic Position

When the mouse is located over a mic stand, the mouse pointer changes to an "updown" arrow and the mic gets a copper/goldish glow. Click-and-drag the mouse up or down to change the position of the mic stand. The microphone moves along a predetermined path, so you only need to move the mouse up or down, not to the left or right.

Cabinet Selection

You can change the cabinet by clicking on the background and dragging the mouse to the left or right. The mouse pointer becomes a "left-right"arrow when you are pointing at the background to indicate that it is possible to change cabinet.

> Clicking on the background while holding the Shift-key toggles through the cabs without any sliding animations.

Mix Panel (Bottom Area)

In the mix panel, you can change the balance between the amplifier signal and the D.I. signal. The tonality of the D.I. signal can be changed with the three EQ knobs and the compressor/limiter. You can also change the phase of the amplifier signal and adjust the output volume.

Amp Panel

Room View

DI and Mix Panel

1. High/Low
2. Normal/Volume
3. Bass
4. Middle
5. Treble
6. Master volume

The Amplifier

The amplifier that we chose to model is actually a guitar amplifier, but because of its very characteristic "non-fizzy" distortion and specialsounding tone stack, it's been very popular among bass players during the last decades. When the **High/Low** switch is set to HIGH, the amplifier will work and act just like the real thing. The LOW mode is the same amplifier with a slightly different input stage, one which lowers the input gain a little bit while adding some warmth to the signal. The LOW mode gives you a fat bass sound without too much distortion.

Use the two "volume" controls as a means of dialing in the right amount of distortion, not to change the actual volume. More **Normal Volume** gives you greater preamp distortion, and more **Master Volume** gives you greater power amp distortion.

High/Low Use this switch to select either the attenuated channel (LOW) or the regular channel (HIGH).

Normal Volume This knob controls the amount of preamp distortion, and is usually called "Gain" on more modern amps. Use it to dial in the amount of distortion you want.

Bass, Middle, Treble These knobs are the tone controls of the amplifier. A common setting is **Bass** on full, **Middle** on min, and **Treble** somewhere in between.

Master Volume Adjusts the amount of power amp distortion. With a massive amount of **Master** and **Normal Volume**, the amp will start sound like it's about to break. Unless you really want that kind of distortion you'll find that a **Master Volume** at 12 o'clock will suit most of your needs.

Master Volume won't distort much unless you increase the **Normal Volume**. Just like the real amp.

The Cabinets

Click and drag left/right on the background to change cabinets.

8x10

The industry standard 8x10" doesn't need much presentation. With a fat sound and lots of low end, this is the goto cabinet for many bass players.

4x12

The regular 4x12" cabinet is preferred by many bass players over the industry standard 8x10". It has a more focused sound than the 8x10", and although it's a bit thinner than the 8x10", it is often easier to work with in a mix.

1x12

This is the odd bird among the cabinets. While trying out cabinets for Bass Amp Room, we felt that we needed to listen to something completely different just to clear our minds. We found this old openback 1x12", plugged it in, and immediately fell in love with the sound. Some sort of '60s sound with lots of room. And you can dial in more low end if you need it just by moving the mic more to the off axis position.

Positioning the Mic

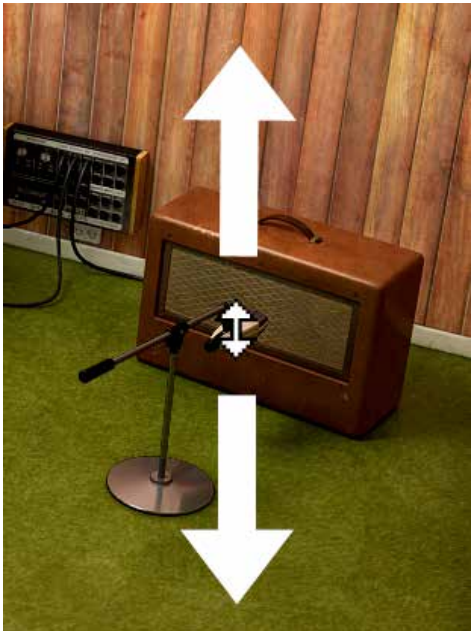
By changing the mic position, you can get lots of different sounds from a single cabinet. How the sound changes depends on the cabinet and the room, but there are some common features for all the cabs in Bass Amp Room:

CLOSE OFF AXIS: The position with most bass and the least amount of highs.

CLOSE ON AXIS: Moving the mic towards the onaxis position will gradually give you more highs and a more pronounced mid. Usually the most "focused" sound.

FAR FIELD: Moving away from the cabinet will give you more room ambiance and often (but not always) a thinner sound.

> If you find that the selection of cabinets isn't enough for you, you can always bypass the cabinet in Bass Amp Room and use the cabinets from one of our other Amp Room plugins (with the amp modeling bypassed).



1. Low Cut
2. Tone
3. High Cut
4. Limiter
5. DI/Amp balance
6. Output volume
7. Solo DI
8. Phase invert (Amp)
9. Solo Amp

The Mix Panel

SO, WHAT ELSE DO YOU NEED EXCEPT a good sounding amp? Since a lot of engineers and producers prefer to work with both the raw D.I. signal and the mic:ed signal (using the amp signal for character and the D.I. signal for focus or low end), we decided to incorporate that work flow into Bass Amp Room. But instead of having two tracks (one with the microphone signal and one with the direct bass line), you can easily blend the two signals directly in Bass Amp Room. And, to make life easier, we added a couple of very powerful tone controls and a stateoftheart singleknob compressor in the D.I. section. That way you will be able to get a really good bass sound just by using the D.I!

Direct Inject (D.I.) Knobs

Low Cut A 12dB/octave low cut filter. This control is intended for filtering out *unwanted* frequencies, but can, of course, be used more creatively. Keep increasing the knob until you cannot hear any difference in the sound. When you start to lose the low end, stop and go back a couple of millimeters (yes, we are metric).

High Cut Same as the Low Cut, but for high frequencies.

Tone When this knob is set at 12 o'clock (FLAT), it doesn't change the sound at all. Turning it clockwise will give you a fatter and more scooped sound much like the tone stack in the amplifier. Turning it counter-clockwise will give you a sound with lots of mid.

Changing the **Low Cut** and **High Cut** will drastically change the characteristics of the **Tone** knob. Technically speaking, the center frequencies of the **Tone** knob filters will change when you change the bandwidth (the settings of **Low** and **High Cut** combined).

Limiter Use the limiter to increase the amount of compression. This function can, for instance, be used to limit the peaks of the D.I. signal, and would then be set somewhere around 12 o'clock, depending on the level of the input signal. With a higher setting you will get a longer sustain on each note.

Limiter LED The LED on the top right of the **Limiter** knob indicates when the limiter is working (like a gain reduction meter).

Mix Knobs

D.I./Amp Balance This fader sets the balance between the D.I. signal and the amp signal. Use the **Solo** knobs to audition each channel (post fader).

Solo Click the Solo texts to solo either the Direct Inject channel or the Amplifier (and cabinet) channel. You can only solo one channel at a time, so by soloing one channel you automatically un-solo the other channel.

Phase Invert (Amp) Inverts the phase of the amp signal. Changing the relative phase between the D.I. signal and amp signal will often completely change the sound. If you have a hard time trying to find a good blend between the two signals, try flipping the **Phase Invert**. It usually makes a big difference to the sound.

Phase invert! If you have the balance fader somewhere in the middle, you should make it a habit to toggle the **Phase Invert** switch. It can make a huge difference to the sound.

Output Sets the output volume. It goes from 32 dB to +32 dB, except at the minimum position where it turns off the output signal completely.

Output Peak LED The Output Peak LED indicates when the output signal peaks at 0 dBFS. Bass Amp Room will not clip the peaks, it is only an indication.

Bypassing Amps or Cabs

You can choose to bypass the amp or the cabinet by selecting AMP BYPASS or CAB BYPASS from the small box in the lower right corner.

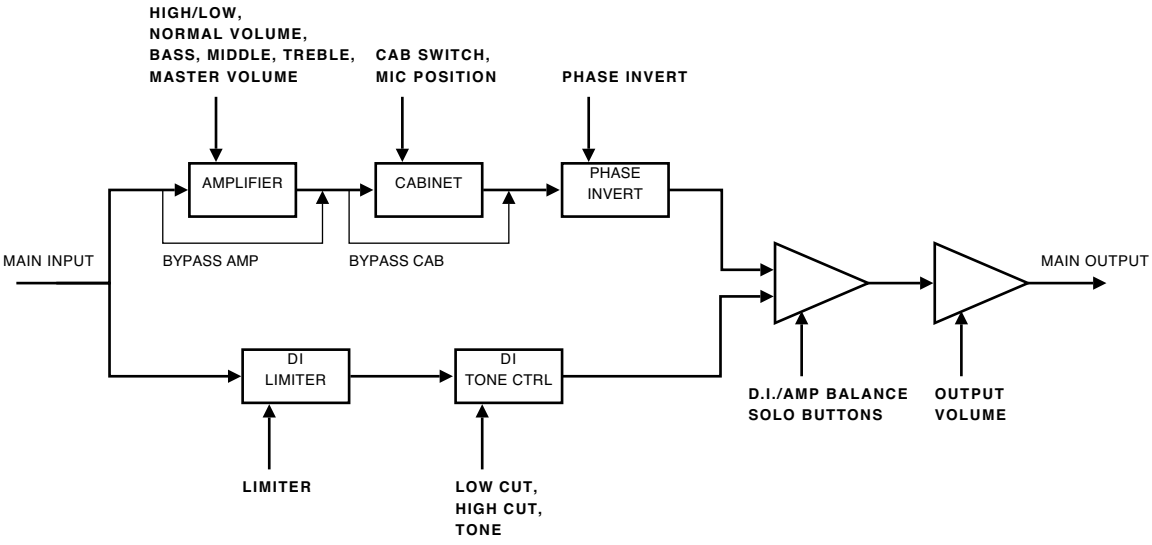
This is very useful if you want to use Bass Amp Room with an external speaker (or speaker plugin) or if you just want to add a cabinet simulation to a track, such as a recorded line out from your amplifier.

This will not bypass the D.I. section, since that can be easily done by setting the **D.I./Amplifier Fader** all the way to the right (on AMPLIFIER).

By bypassing the cabinets in Bass Amp Room, you can use the cabinets from the other Amp Room plugins. Just insert, for example, Metal Amp Room (with bypassed amplifier) on the track after Bass Amp Room (with a bypassed cabinet).

Block Diagram

The bass signal (Main Input) is split up into two identical signals, one that goes to the amp (top section) and one to the D.I. (bottom section). The two signals are mixed with the D.I./Amp Balance fader.



Buying Recommendations

If you like the sound from Bass Amp Room and would like to get that sound using the real deal (let's face it, a real amp is always sexier than a plug-in), here are some buying recommendations.

Amplifier

If you like this characteristic fat distortion and scooped-out tone stack, I would suggest that you get a two-channel Hiwatt DR103 from the late '70s/early '80s.

Cabinets and Mic

The leftmost cabinet was modeled after the industry standard of bass cabinets: the Ampeg 8x10" SVT810. If you like the 4x12 sound, I'd suggest that you try out a couple of different 4x12s, because we have absolutely no clue what kind of 4x12 we measured. It sounded excellent, had no labels on, and wasn't for sale... The small cabinet was a half-open 1x12" cabinet with a Celestion 20W/15ohm driver.

But you will also need a good sounding mic to get a good bass sound, and in our opinion nothing beats the Sennheiser MD421!

D.I. and Mix Panel

The tone controls of the D.I. and limiter was made by us and have no hardware counterpart.

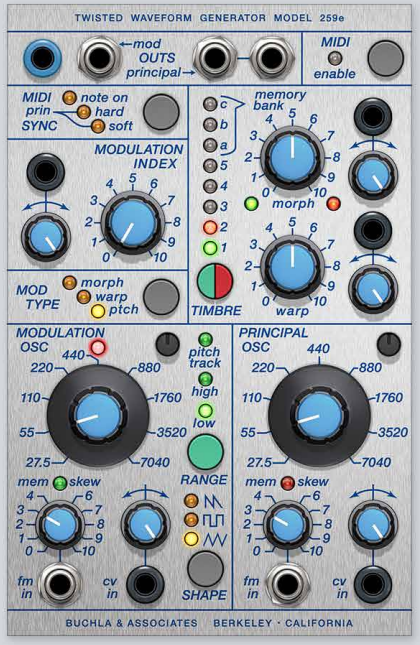
Room

If you like the sound of the room, make sure to schedule a session at Care Of Sound in Norrköping.

BRAND NAMES ARE TRADEMARKS OR REGISTERED TRADEMARKS OF THEIR RESPECTIVE OWNERS.

Credits

Niklas Odelholm – modeling and sound design. **Oscar Öberg** – compressor modeling, framework programming. **Torsten Gatu** – interface programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering.



MODULAR EXCLUSIVE

10 Buchla 259e Twisted Waveform Generator

Donald Buchla, an inventor based in Berkeley, USA, started building and designing electronic instruments in the early 1960s. It all started when he was commissioned by avant-garde composer **Morton Subotnick** to build a modular instrument for composing and performing live electronic music. The result was called the **Buchla Box**—Don Buchla's first modular synthesizer. This evolved into Don's first commercially available system, the Buchla 100 system, which was followed in the 70s by the legendary **Buchla 200 system**.

In the early 2000s, Don decided to reinvent and reimagine the 200-system, and the new Buchla 200e system was born. The 200e system is an interesting hybrid between analog and digital technology, and the Buchla 259e module is a perfect example.

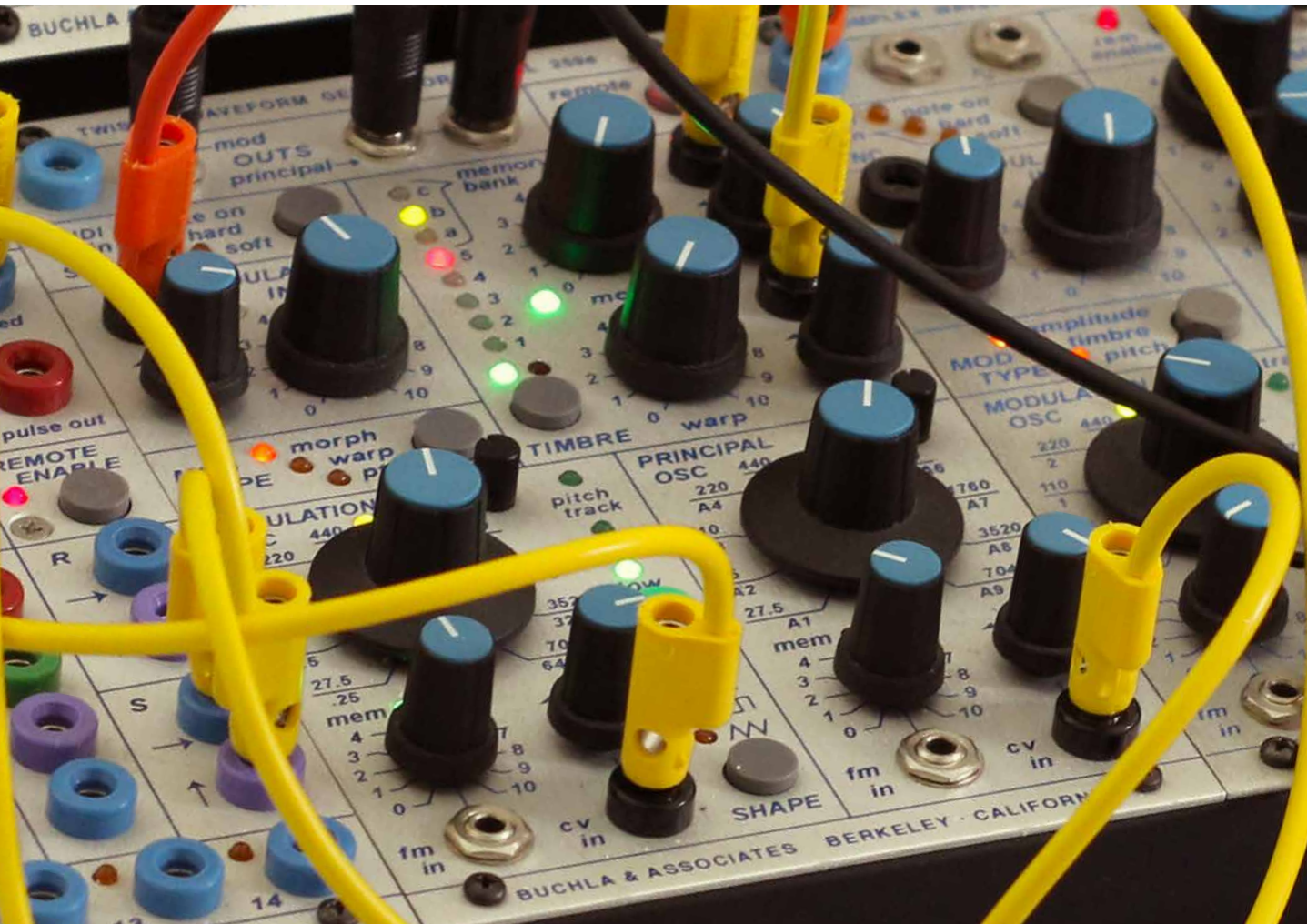
The **259e** is in one sense a modern day digital implementation of the original **Buchla 259 Complex Wave Oscillator**, but in many other ways it's a different beast altogether. The 259e's crazy memory scanning functionality and its cold and very digital sounding waveshaping capabilities, with plenty of characteristic aliasing distortion, makes it a prime candidate for sonic experimentation beyond the ordinary.

Overview

Softube's **Buchla 259e Twisted Waveform Generator** plug-in for **Modular** features the digital waveshaping, aliasing noise and fold over frequencies of the original hardware. The plug-in captures all the self-modifying, screeching, snarling responses from the original's downright odd inner workings—which added "twisted" to its name.

The 259e consists of two separate oscillators. One is the Modulation oscillator, which primarily serves to modulate the second one, called the Principal oscilla-

The Buchla 259e hardware



tor. The Modulation oscillator can be switched between High and Low frequency ranges, where High generates notes in the audible region and Low is mostly for when the oscillator is used as an LFO which modulates the Principal oscillator. The Modulation oscillator can be switched between three classic waveform shapes, and three different modulation targets can be selected, one by one or in any combination.

The sine wave generated by the Principal oscillator is simultaneously applied to two of the eight available waveshape tables. Step through the tables (1-5, a-c) by pushing the green and red buttons. Green and red LEDs indicate which tables are currently selected.

Tables a, b and c are actually not tables in the classical sense—they are simply portions of the 259e operating program, full of unpredictable noise and frequent silences. When these tables are selected, the FM controls are re-assigned to table scanning functions and the FM inputs become table modulators. In this state, the Mem Skew LEDs above the FM controls light up.

A morph voltage pans between the two tables and a warp voltage varies the amplitude of the sinusoidal (driving) waveform. Both these functions can be modulated by the modulation oscillator.

Buchla Philosophy

The Buchla systems vary the harmonic content in their patches by using a waveshaper on a rudimentary sound source such as a sine oscillator, rather than shaping of harmonic content through filtering, which is the more commonly used method. Buchla's type of synthesis is often referred to as the West Coast style.

Where a VCF removes harmonic content from a signal, a waveshaper warps the waveform by applying either a fixed or variable mathematical function. This real time process alters the shape of the waveform, adding harmonics and resulting in a more abrasive sound compared to the typical East Coast style characteristics (classic subtractive synthesis).

In a Buchla synthesizer system, audio and modulation signal paths are separated by their different jack and cable formats. Audio jacks are silver and AC coupled, while modulation jacks are colored "banana" jacks which are DC coupled. The form factor of the jacks and the different cables make it easy to see which signal goes where and creates a natural signal flow in the Buchla environment.

Softube has reimagined the 259e in a Eurorack environment, adapting levels and connectivity to work as any Eurorack module. This means that the 259e module featured in Modular, unlike its hardware counterpart, will have no limitations when it comes to connectivity—you can connect any type of output to any kind of input, whether it's audio or modulation. But for the sake of familiarity, we kept the look and color of the hardware audio and modulation jacks. Softube has also adapted the signal levels to harmonize better with the other modules in the Modular system.

The following adaptations have been made when converting the Buchla 259e to Softube Modular:

* Audio and CV can be freely patched to each other.

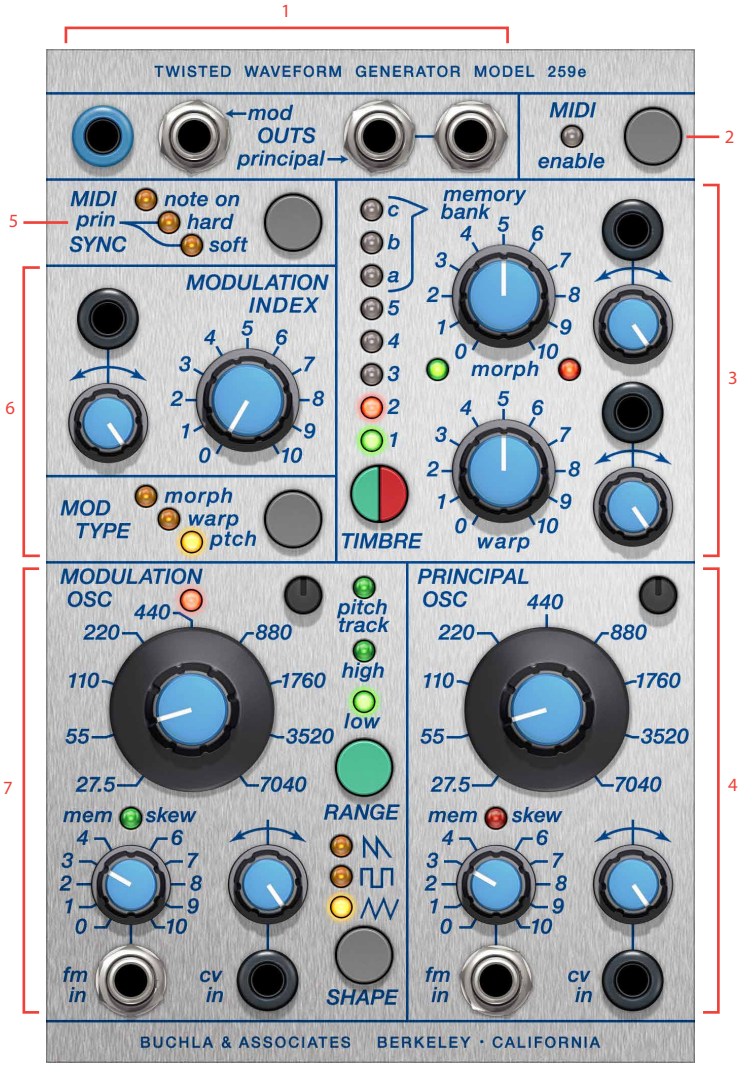
* CV voltages rescaled by around 0.9 times, so that setting the frequency CV attenuators to max gives a 1V/oct response.

* Audio voltages have been rescaled by around four times, to match Eurorack levels better.

* Remote Enable replaced by MIDI Enable. Press once to go to MIDI learn mode (blink), play a MIDI note and it locks to that channel. Press again to turn MIDI off.

* the AC coupling of the FM jacks was disabled when in Slew mode in order to make them more useful when in use with regular DC CV modulation such as LFOs, envelopes and sequencers.

Overview



1. Main outputs
2. Midi enable
3. Principal oscillator Timbre section
4. Principal Oscillator
5. Modulation oscillator Sync behaviour
6. Modulation conversion between Modulation oscillator and Principal oscillator
7. Modulation oscillator

Detailed Description

MIDI Enable Button

When clicked once in its preset state, the LED will start flashing, an indication for that the 259e is awaiting a MIDI note on message. When the MIDI note is received, the 259e automatically sense will sense on which MIDI channel and lock onto that channel (LED is lit). By clicking once more, MIDI is reset to off status and LED will go unlit. Received MIDI range is from A-1 to A7 (8 octaves), same range as coarse tune. MIDI pitchbend is received on the same MIDI channel and responds to +-1 octave.

MIDI Prin Sync Button (Modulation Osc Sync)

This button toggles between the different reset behaviors of the Modulation Oscillator:

- **Preset mode** - no synchronization is set (all LEDs are unlit).
- **Soft sync mode** - uses a phase-lock loop to achieve synchronization.
- **Hard sync mode** - truncate synchronization to the Principal oscillator. This means that the Principal oscillator will hard reset the Modulation oscillator.
- **MIDI note sync** - synchronisation by incoming MIDI note messages on the channel set by the MIDI enable function (see description above). This means that this function will not be operational when MIDI enable is set to off.

Morph

The **Morph** knob is the offset used to fade between the green and red wave-shape tables of the Principal oscillator. When this knob is fully counter-clockwise and no modulation is applied to morph from neither the Modulation oscillator or the **Principal Osc Morph CV** jack, the green LED just beneath the knob will be lit indicating that presently, only the green wave-shape is output on the Principal outputs. When turning the morph knob gradually clockwise, more and more of the red wave-shape will be faded into the Principal outputs, and at fully clockwise, only red is output. The **Morph** knob works as an offset, to which modulation is added, either via the **Principal Osc Morph CV** input or via the **Modulation Index** routing.

Principal Osc Morph CV

This is the bi-polar amount knob to scale the incoming CV at the **Principal Osc Morph CV** jack. At 12 o'clock position, amount is 0 and then increasing clockwise up to 1 times scaling at the 5 o'clock position. Turning the knob counter-clockwise from 12 o'clock to 7 o'clock position sets amount to fully -1 times (CV has inverted impact on morph). Modulating morph CV is always relative to the position of the **Morph** knob which works as an offset.

Modulation Index CV

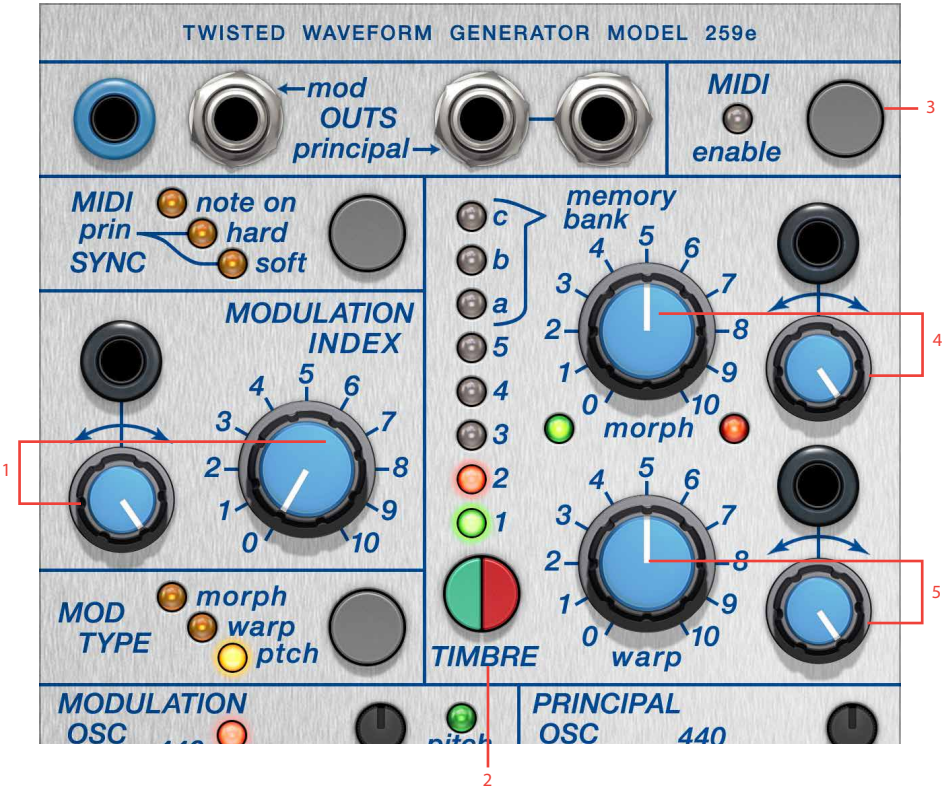
This is the bi-polar amount knob to scale the incoming CV at the **Modulation Index CV** jack. At 12 o'clock position, amount is 0 and then increasing clockwise up to 1 times scaling at the 5 o'clock position. Turning the knob counter-clockwise from 12 o'clock to 7 o'clock position sets amount to fully -1 times (CV has inverted impact on the Modulation Index). Modulation CV is always relative to the position of the Modulation Index knob (offset). See further description on Modulation Index below. The signal patched into the **Modulation Index CV** input will affect the amount of modulation selected via Mod Type and offset via the Modulation Index, positively or negatively. For example FM amount between Modulation oscillator and Principal Oscillator can thus be affected by an external ADSR envelope.

Modulation Index

The **Modulation Index** knob sets the amount of the selected modulation between the Modulation oscillator and the Principal oscillator. This is done via Mod Type button (see description below). The **Modulation Index** knob works as an amount offset, which is affected by the signal at Modulation Index CV input. This signal can be set to affect the Modulation Index positively or negatively.

Timbre Bank Buttons

These green and red buttons control the different wave-shape tables of the Principal oscillator. There are different wave-shape tables for the green and red sides, although some of them are the same. When clicking ov n one of the green and red buttons, the corresponding colored LED will move one step up the wave-shape selector ladder. Both sides, green and red, will share the same indicator which makes the LED lit up orange when green and red indication appears on the same step. When clicked through until step C, one more click will cause the that side to cycle back to step 1. Green and red **Timbre** banks are the same for position 1, A, B and C, but differ slightly in harmonics for position 2,3,4 and 5 on the green and red side. Positions 1 through 5 are different regular wave-shapes, while position A-C reflect the fun and quirky memory bank mode. In this mode, the wave-shapes are based on scanning through the digital code memory of the Buchla 259e with the **Mem Skew** function (see further description below).



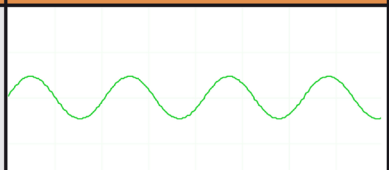
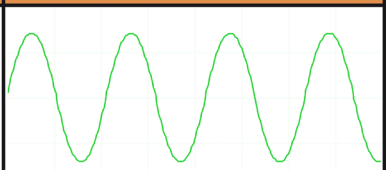
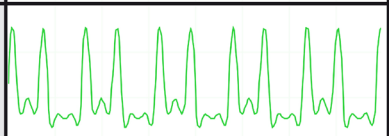
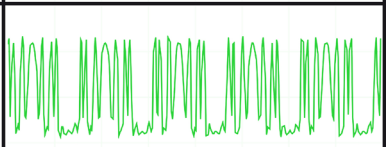
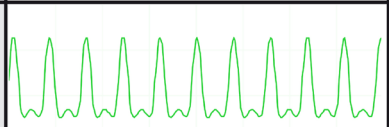
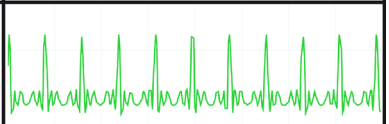
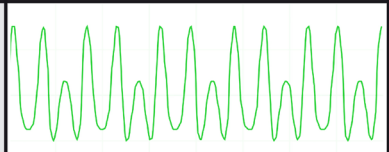
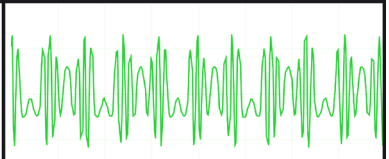
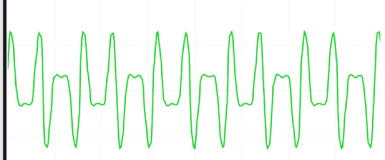
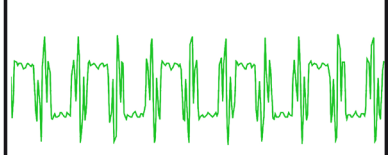
1. Modulation Index Offset and Attenuator
2. Green and red Timbre selection
3. Midi enable button
4. Morph knobs
5. Warp knobs

1. Modulation oscillator coarse and Fine tune
2. Green and red FM Amount (or Mem Skew)
3. Pricipal Coarse and Fine tune

Principal Osc Warp

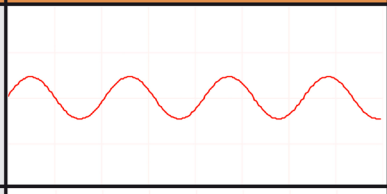
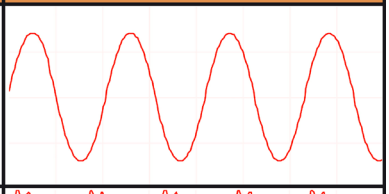
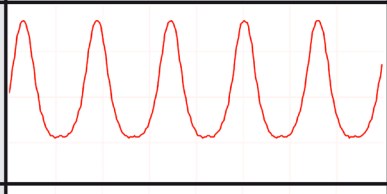
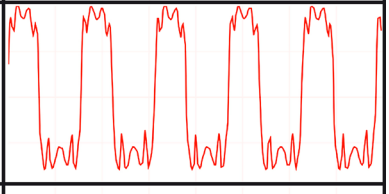
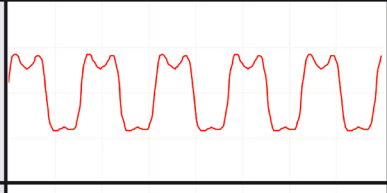
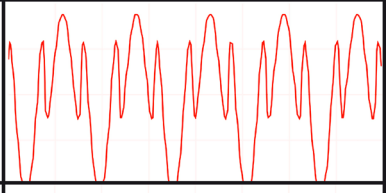
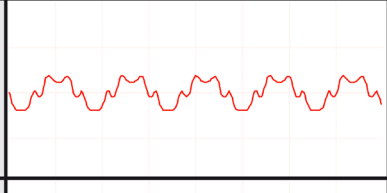
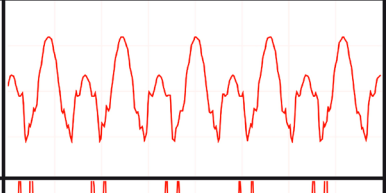
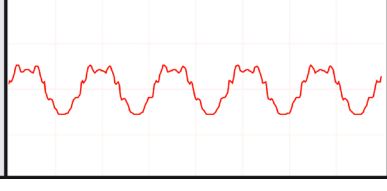
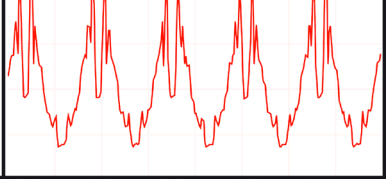
The **Osc Warp** knob controls the folding of each of the timbres (except timbre nr 1, sine) working as an offset for the modulation applied to the Principal **Oscillator Warp CV** input. Increased warp changes the output harmonics of the Principal Oscillator. When the in **Mem Skew** mode, the **Principal Osc Warp knob** will set the range of Memory being scanned via the **Mem Skew Index** and CV input.

Buchla 259e Principal Oscillator Timbres: Green Timbres

Timbre	Warp = 0	Warp = 10
Bank 1		
Bank 2		
Bank 3		
Bank 4		
Bank 5		

Waveforms captured at 440hz

Buchla 259e Principal Oscillator Timbres: Red Timbres

Timbre	Warp = 0	Warp = 10
Bank 1		
Bank 2		
Bank 3		
Bank 4		
Bank 5		

Waveforms captured at 440hz

Principal Osc Warp CV

This bi-polar amount knob to scales the incoming CV at the **Principal Osc Warp CV** jack. At 12 o'clock position, amount is 0 and then increasing clockwise up to 1 times scaling at the 5 o'clock position. Turning the knob counter-clockwise from 12 o'clock to 7 o'clock position sets amount to fully -1 times (CV will have inverted impact on the **Principal Osc Warp)**. **Principal Osc Warp CV** is always relative to the position of the main **Principal Osc Warp knob** (offset). The signal patched into the Principal Osc Warp CV input will affect the Warp amount, positively or negatively.

Mod Type button

This button determines which destination, or combination of destinations, the Modulation Oscillator will modulate through the Modulation Index. Multiple presses on this button switches between the none (all three LEDs - morph, warp and ptch, are unlit), one, two or all three of the available Modulation destinations: **Morph, Warp** and **Ptch** (Pitch). There are in all 7 different combinations available

Modulation Osc Fine

This knob is for fine-tuning the pitch of the Modulation Oscillator. The range is roughly up and down two semitones.

Principal Osc Fine

This knob is for fine-tuning the pitch of the Principal Oscillator. The range is roughly up and down one and a half semitone.

Modulation Osc Coarse

The **Modulation Osc Coarse** knob sets the coarse tuning for the Modulation oscillator. When **Modulation Osc Range** is set to Low the range of this knob goes from 0.25Hz to 64Hz. When set to High range is 27,5Hz to 7040Hz. In Pitch Track mode, the knob function as offset to pitch set by the Principal Oscillator frequency.

Principal Osc Coarse

The **Principal Osc Coarse** knob will set the coarse tuning of the Principal oscillator ranging from 27,5Hz up to 7040Hz. In MIDI mode this knob will not always reflect the pitch heard because, in MIDI mode, pitch is determined by the last MIDI note played. And when you then tweak the Coarse-knob, it will work as an offset from the MIDI note. In practise this means that if you play an A4 in MIDI mode, the following time you adjust the **Principal Osc Coarse** knob, the pitch of the Principal Oscillator will start at 440Hz regardless of where the knob points, and work in linear fashion towards the lo and hi extreme endpoints of the knob. A new MIDI note will force the frequency back to that note value. This means that if a MIDI sequence is played, the **Principal Osc Coarse** knob can be used to temporarily tweak the frequency of the active notes, while it will always jump back in tune for every received MIDI note.

Modulation Osc Range button

- **Pitch track** - when this LED is lit, the Modulation Oscillator pitch is tracking that of the Principal Oscillator, but offset by the **Modulation Osc Coarse** knob. In this mode, the LED above the **Modulation Osc Coarse** knob indicates if the pitch-relation to the Principal Oscillator is below (red light), above (green light), or spot on (LED is unlit) when the **Modulation Osc Coarse** knob is set to 12 o'clock.
- **High** - the Modulation Oscillator operates in the audio-region with the **Coarse** knob ranging from 27 to 7040 Hz. In this mode, the LED indicator above the **Coarse** knob remains unlit
- **Low** - the Modulation Oscillator operates in the sub-audio range with the **Coarse** knob ranging from .25 Hz to 64 Hz. The LED indicator above the **Coarse** knob in this mode reflects the cycles of the Modulation Oscillator (red being negative part and green being positive part).

Modulation Osc FM

This knob sets the amount of frequency modulation to affect the Modulation Oscillator when patching in an external signal into the FM in input jack. When the Principal Oscillator's green timbre table is in **Mem Skew** mode (using bank A,B or C), this knob's functionality is re-assigned to table-scanning. This is indicated by the **Mem Skew** green LED indicator being lit above this knob. In Mem Skew mode, the **Modulation Osc FM** knob will act as an offset for the Principal Oscillator green table scanning while signals patched into the FM in input jack will affect the scanning (without any attenuation).

Modulation Osc CV

This knob will attenuate modulation signals patched into the CV in input jack situated below this knob. Knob position 12 to 5 o'clock will attenuate positively clockwise (0 to 1 v/octave), while knob position 12 to 7 o'clock will attenuate negatively anticlockwise (0 to -1v/octave).

Principal Osc FM

This is the amount of frequency modulation to affect the Principal Oscillator when patching in an external signal into its' FM in input jack. When the Principal Oscillator's red timbre table is in **Mem Skew** mode (using bank A,B or C), this knob's function is re-assigned to table-scanning. This is clearly indicated by the **Mem Skew** red LED indicator being lit above this knob. In **Mem Skew** mode, the Principal Osc FM knob will act as an offset for the Principal Oscillator red table scanning while signals patched into the Principal Oscillators FM in input jack also will affect the scanning (without any attenuation).

Principal Osc CV

This knob will attenuate modulation signals patched into the **Principal Oscillator CV** in input jack situated below. Knob position 12 to 5 o'clock will attenuate positively clockwise (0 to 1 v/octave), while knob position 12 to 7 o'clock will attenuate negatively anticlockwise (0 to -1v/octave).

Indicators

MIDI enable LED

This LED starts blinking when clicking on the MIDI mode button, indicating that the 259e is waiting for a MIDI note on message to determine which MIDI channel to listen to. When having received a MIDI note, this LED will stop blinking and remain lit until MIDI mode button is clicked again to switch 259e to manual mode.

MIDI prin sync LEDs

This LED indicates sync status of the Modulation Oscillator. For further detailed information on the different indicated sync modes, see description above (MIDI prin sync button).

Timbre Bank LEDs

These LEDs tells you which timbre banks, green and red that are selected for the Principal Oscillator.

Morph LEDs

These LEDs indicate the balance of the timbre banks heard on the Principal Oscillator output, directly reflecting the sum of the **Morph** knob and **Principal Osc Morph CV** input.

Mod Type LEDs

These LEDs indicate which modulation destinations in the Principal Oscillator that currently are selected to be connected to the Modulation Oscillator. Selected destinations can be any combination of one, two or all three destinations. None is also an option (no LEDs lit).

Modulation Osc LED

Situated just above the **Modulation Osc Coarse** tune knob is the Modulation Osc LED reflecting the internal function of the oscillator. When the Modulation Oscillator is set to Low Mode it will reflect the oscillator rate with red being the negative portion and green being the positive portion of the waveform.

Mem Skew LED Green
This LED is lit when the Principal Oscillator green timbre bank is using one of the memory banks A, B or C. It indicates the temporarily changed functionality of the **Modulation Osc FM** knob and CV input (for further detailed information, see above description about the **Modulation Osc FM** knob).

Modulation Range LEDs
These LEDs reflect Modulation Oscillator operational mode: **Low, High** or **Pitch Track**. See further detailed description above on the **Modulation Oscillator Range** button.

Mem Skew LED Red
This LED is lit when the Principal Oscillator red timbre bank is using one of the memory banks A, B or C. It indicates the temporarily changed functionality of the **Principal Osc FM** knob and CV input (for further detailed information, see above description about the **Principal Osc FM** knob).

Inputs

Principal Osc Morph CV input
This is the CV input for the blend of the red and green timbre banks on the Principal Oscillator.

Modulation Index CV input
CV input for scaling the Modulation Index between the Modulation Oscillator and Principal Oscillator.

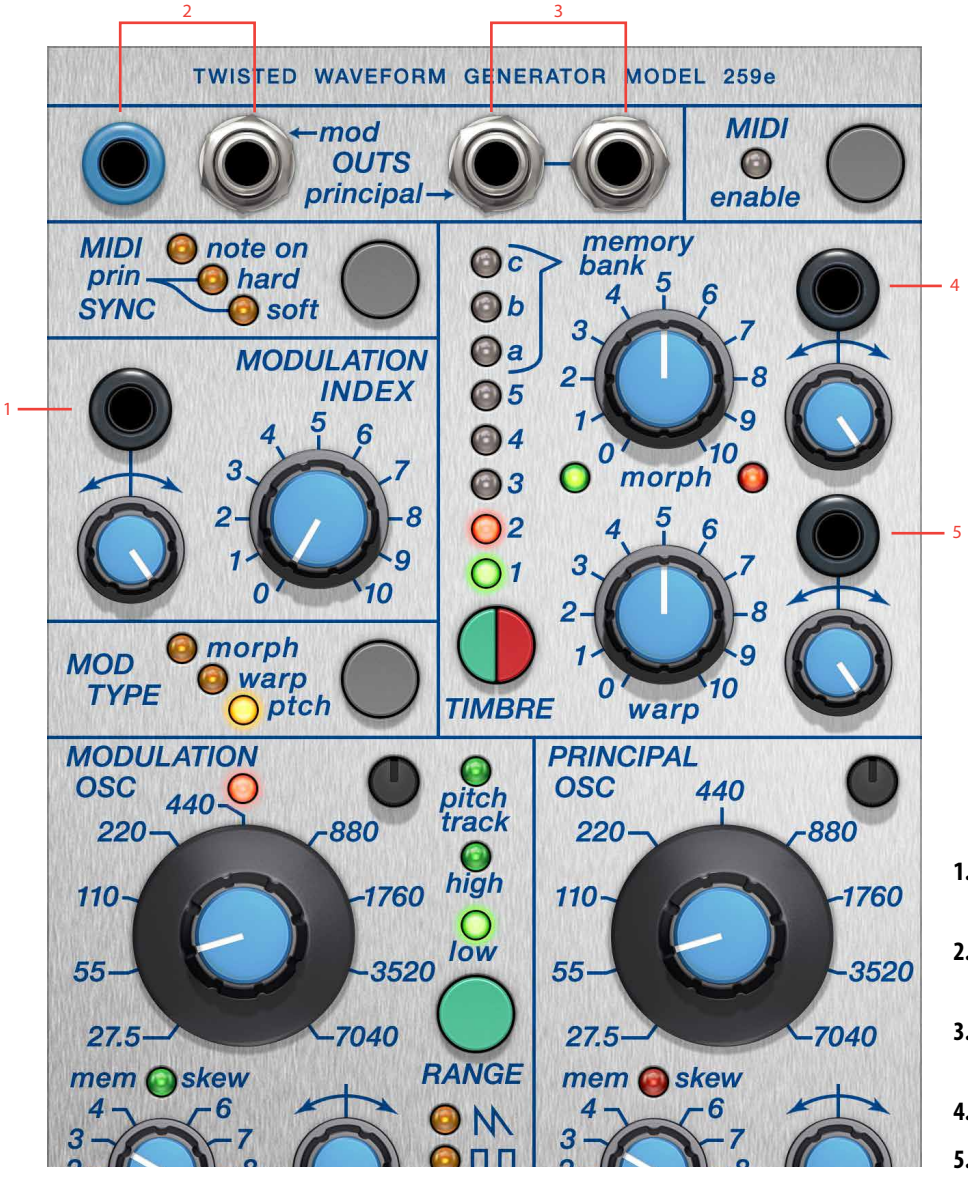
Principal Osc Warp CV input
This jack is the input for externally controlling the **Warp** function of the Principal Oscillator from a CV modulator (LFO, envelope, etc).

Modulation Osc FM input
Audio input for frequency modulation of the Modulation Oscillator or, in **Mem Skew** mode, external CV control of the green timbre banks memory scanning. This jack is AC coupled in ordinary FM mode which means that it will work better with audio signals (this is how the original hardware works). In **Mem Skew** mode this jack will change to DC coupled mode to better work with modulation signals such as LFOs, envelopes and sequencers.

Modulation Osc CV input
This jack is external CV control of the Modulation Oscillator pitch, tracking 1V/octave when the **Modulation Osc CV** knob is set to max (preset value).

Principal Osc FM input
Audio input for frequency modulation of the Principal Oscillator or, in **Mem Skew** mode, external CV control of the red timbre bank's memory scanning. This jack is AC coupled in ordinary FM mode which means that it will work better with audio signals (this is how the original hardware works). In Mem Skew mode this jack this jack will change to DC coupled mode to better work with modulation signals such as LFOs, envelopes and sequencers.

Principal Osc CV Input
This jack is external CV control of the Principal Oscillator pitch, tracking 1V/octave when the Modulation Osc CV knob is set to max (preset value.



1. Modulation attenuation CV in
2. Modulation oscillator CV and Audio output
3. Principal oscillator outputs
4. Morph CV in
5. Warp CV in

1. Modulation oscillator
   FM and CV in
2. Principal audio
   FM and CV in

Outputs

Modulation Osc CV Out

This is the DC coupled output of the Modulation Oscillator spanning approximately 0 to 8.6V. This output is very suitable when using the Modulation Oscillator as an LFO to modulate other module.

Modulation Osc Audio Out

This is the AC coupled output of the Modulation Oscillator. Use it when you're patching the Modulation Oscillator as an ordinary oscillator to external filters and such.

Principal Osc Audio Dual Out

These two identical AC coupled outputs are the main outputs of the Buchla 259e where the resulting waveform of the Principal Oscillator is heard.

The Buchla 259e Module In Use

Due to its complexity, patching the 259e can be pretty challenging but also great fun.

Here are some interesting patch-scenarios for you to try out:

Building Filter-Like Effects With 259e

Morphing between a timbre bank with little harmonics and a timbre bank with lot of harmonics can give the you the illusion of a filtered sweep. Try for example using green timbre bank 1 and red timbre bank 3 for a sine to square like sweep. Experiment with different warp amounts to give the red timbre part different harmonic content.

Synced Modulation

The fact that the Modulation Oscillator can be synced to incoming MIDI note messages and slaved to the Principal Oscillators pitch makes it interesting for audio rate modulation of pitch, Morph and Warp. Try for instance to have the Modulation oscillator to rapidly Morph the Principal Oscillator banks while in pitch track mode.

Use a envelope generator gated from a MIDI to CV module to sweep the Modulation Index and this will create a chorusing motion in the harmonics of the Principal Oscillator output.

Modulation Oscillator As External LFO

The Modulation oscillator can of course also be used as a ordinary LFO that can control external modules such as another 259e via its modulation (blue jack) and audio (silver jack) outputs. Using the Modulator Oscillator as LFO you can take advantage of the MIDI note on reset if you for instance would like a tremolo effect that will sound exactly the same for each note.

Using Aliasing As A Desirable Effect

The principal oscillator on the 259e has a pretty gritty, nice sound full of gorgeous aliasing in the upper regions. This can be used as a great source of metallic sounding effects, hihats, gongs etcetera

Using The Modulation Oscillator As Main Oscillator With Hard/Soft Sync

The Modulation Oscillator can of course also be used as main audio source. Especially the hard- and soft-sync features add new and exciting harmonic possibilities not offered by the Principal Oscillator. The Buchla 259e sync options do not sound like the other oscillators and have highly musical qualities.

Skew In Practise

The 259e's timbre banks A,B and C enable a special Memory Scan mode where parts of the 259e internal program memory is played back as audio through the Principal Oscillator outputs. In this mode, Warp will set the scope of swept memory and the FM in knob (Mem Skew) will set the offset (i e reference point in memory), where incoming CV 0 to approximately 5.88V will sweep portion of the memory around the set offset. The 259e for Modular has been modified to be used with DC signals which makes it possible to for example very precisely sequence Mem Skew positions with a CV sequencer.

Drones

The dual timbre banks of the Principal Oscillator coupled with the built-in modulation of the Modulation Oscillator makes the 259e ideal for creating long, evolving and exciting drone sounds. Just add a couple of slow LFOs sweeping Morph, Warp, Modulation Index as well as Mem Skew inputs and you'll have a killer drone in no time. Just add reverb.

Creating thundering bass sounds

By mixing the audio outputs of the Modulation Oscillator and the Principal Oscillator and slightly detuning them, the core basis for great bass sounds can easily be created. Use a lowpass filter like the Doepfer A-108 VCF8 or the Doepfer A-101-2 LPG to further filter out high frequencies and get a pronounced bottom end.

All in All: The Buchla 259e is a very creative and musical module that will bring you many hours of pleasant music making. Have fun!

Credits

Oscar Öberg – modeling, project management, validation. **Kristofer Ulfves** – presets, validation, user manual. **Igor Miná** – user manual layout, hardware photos. **Todd Barton** - testing, presets. **Patrick Detampel** – testing, feedback. **Henrik Andersson Vogel** – marketing. **Bitplant** - graphics.

MODULAR EXCLUSIVE

11 Buchla 296e

Donald Buchla, an inventor based in San Francisco, was in the late 1960s commissioned by Avant Garde composer Morton Subotnick to build a modular instrument for composing and performing live electronic music, the Buchla Box - Don's first modular synthesizer. This evolved into Don's first commercially available system, the Buchla 100 system, followed in the 70s by the Buchla 200 system.

The Buchla 296e Spectral Processor of the Buchla 200e system was released in 2010. The 296e has its roots in the classic Buchla 296 Programmable Spectral Processor, part of the original 200-system built in the 1970s and made famous by artists such as Suzanne Ciani and Morton Subotnick. Where the original 296 had more or less the same functions, the 296e utilizes digital technology for entering and presenting information through the touch LED-display.

At the heart of the Buchla 296e are 16 well defined band-pass filters coupled with 16 VCAs and 16 envelope followers. Each band's audio level is under full control from the A-B morph, CV inputs or the Spectrum Xfer (spectral transfer) functionality. There's is also a freeze function for freezing the output level of each band in order to to sampe the spectral "footprint" into one of the A, B Levels as well as into the Decay page memories.

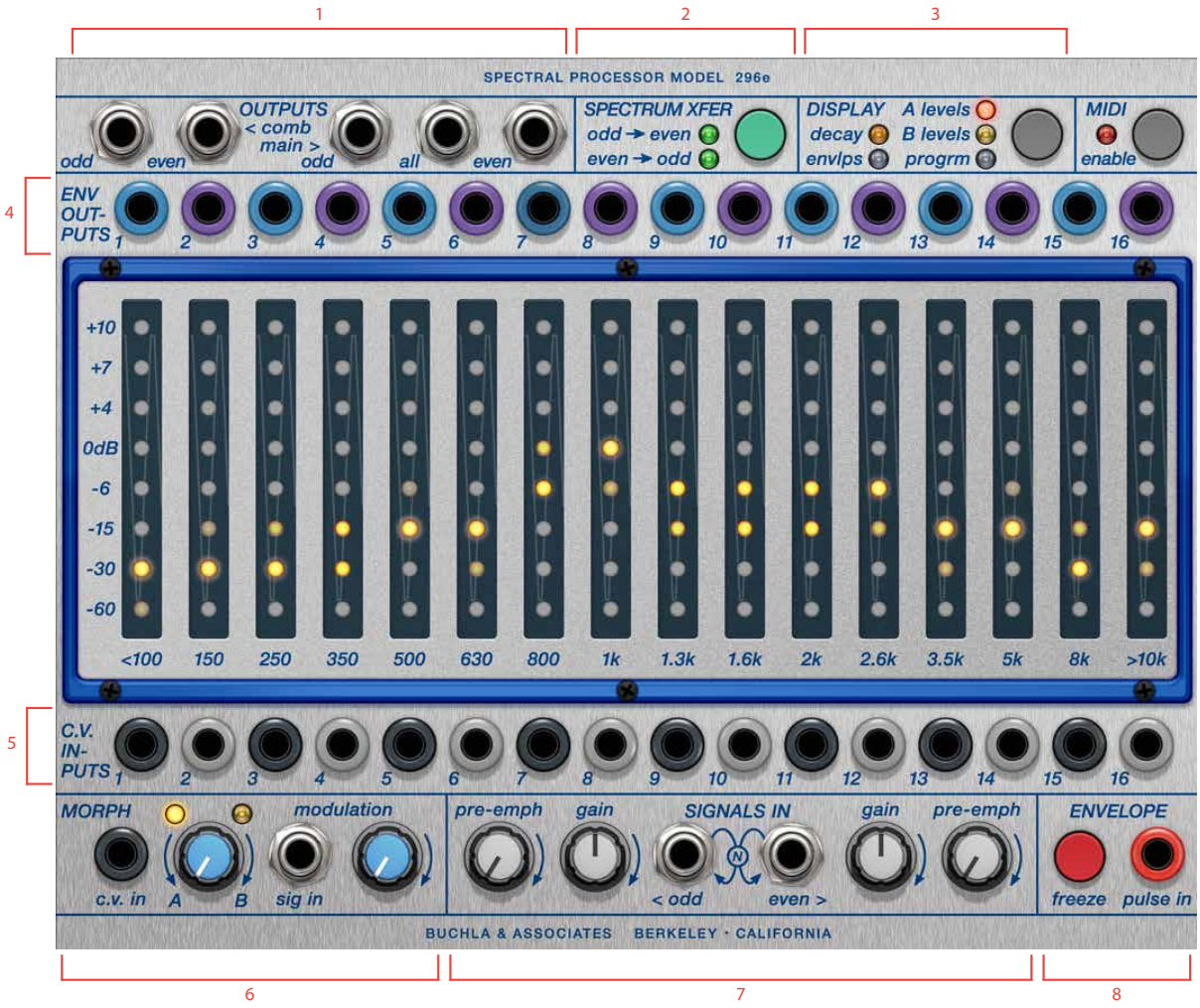
This freeze toggle button feature differs slightly from the momentary switch of the hardware original. There is also a new MIDI mode that replaces the remote button of the hardware that lets you "play" the bands of the 296e from a MIDI keyboard. These differences from the original hardware are described in detail below.

In the "In Use" section at the end of this user manual we have also listed some potential fun use-cases for the Buchla 296e Spectral Processor. Hope you will enjoy the read and have a happy experience using Buchla 296e for Softube Modular.

Buchla philosophy

In a Buchla electronic music system, audio and modulation paths are separated by their different jack and cable formats: Audio jacks are silver while modulation jacks are colored "banana"-jacks which are DC coupled. The form factor of the jacks and the different cables makes it easy to see which signal goes where and creates a natural signal flow in the Buchla environment. Softube has reimagined the Buchla 296e Spectral Processor in the Softube Modular virtual eurorack environment, adapting levels and connectivity for it to work as any other module. This means that the 296e module featured in Modular will have no limitations when it comes to connectivity – you can connect any type of output to any kind of input, whether it be audio or modulation, but for familiarity, we kept the look and color of the hardware audio and modulation jacks.

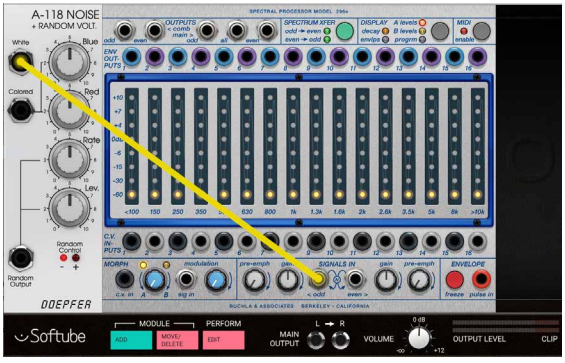
User interface



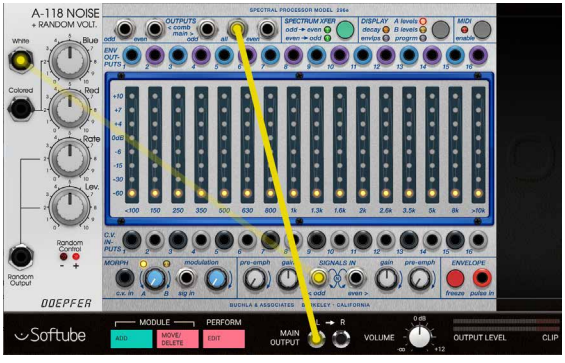
1. Audio outputs
2. Spectrum transfer modes
3. Display modes
4. Envelope CV outputs
5. VCA CV inputs
6. Morph section
7. Audio signal inputs
8. Freeze section

Getting started with Buchla 296e

The Buchla 296e Spectral Processor is a quite complex module but there's an easy way of getting started. Here's how:

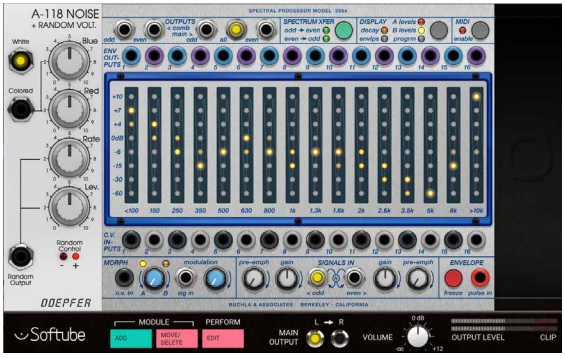


1. Connect a noise-source to one of the signal in jacks at the bottom of the 296e. Make sure that Gain is turned up at least half way.

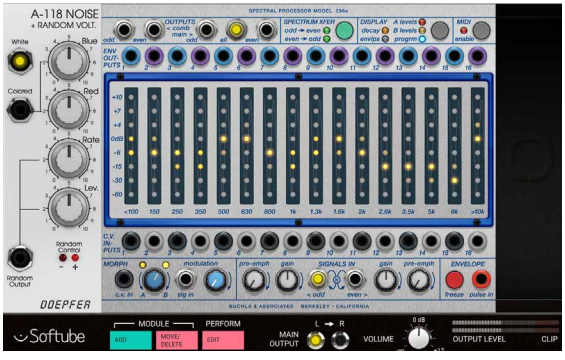


2. Connect the All output at the top of the 296e to Modular's main outputs.

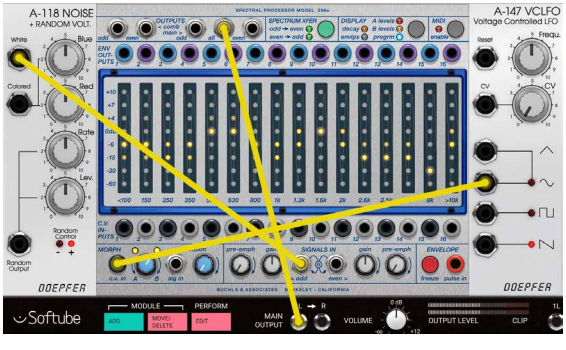
3. Select Display mode A by repeatedly clicking on the Display Levels button until the A Levels LED is lit (orange). Now you can use your mouse-pointer to "draw" a curve across the 16 bands of the 296e. You'll notice that you'll be hearing sound from the corresponding band when its level is turned up.



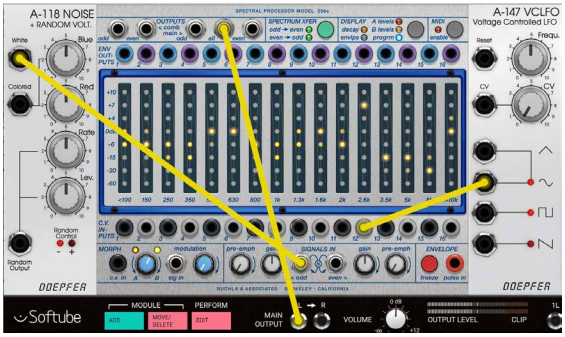
4. Click on the Display Levels button once more and the B Levels LED lights up (yellow). Again, use the mouse-pointer to "draw" a curve across the 16 bands, but this time there won't be any audible difference. This is because we're still listening to the levels set by the A Levels page, indicated by fact that the Morph A-B knob is set fully counter-clockwise (Morph A).



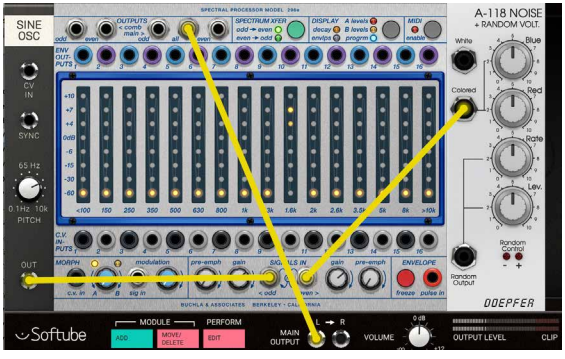
5. Now, click on the Display Levels button once more and the "progrm" option is selected (blue LED lit). Now the LED touch-display will show the resulting output levels. Use the morph A-B knob to "morph" between the A levels and B levels, meaning that output levels adjust seamlessly between the amplitude levels set at the A Levels and B Levels page.



6. Now let's control this motion remotely by connecting a modulator to one of the two modulation inputs. For this purpose I'm using a Doepfer A-147 LFO and connecting it to the Morph CV input. Now, watch as the morph between the A Levels and B Levels are automated. Note that you'll still hear the resulting "programmed output" at the All out audio output although another Display mode is selected and reflected in the touch-display.



7. OK, let's repatch the LFO to control the VCA of a single band instead. By patching the LFO signal into the 2.6k (the 12th band) input, the LFO will add to the signals controlled by the curves drawn on the A and B levels page.



8. Now, let's explore the Spectrum Xfer functionality of the 296e. Xfer meaning transfer and spectrum transfer makes it possible to cross-couple the envelope outputs of the odd (1, 3, 5, 7, 9, 11, 13, 15) to the VCA inputs of the even bands (2, 4, 6, 8, 10, 12, 14, 16) and vice versa.

Let's erase everything in the patch and start again by adding a Sine oscillator and inserting that into the odd input and turning up the gain slightly past 12 o'clock. Now let's add a A-118 noise source and connect it to the even input, turning up the gain also slightly past 12 o'clock. So far you will hear both sources (the sine-oscillator and noise) from the All output of the 296e.

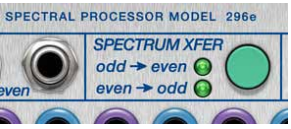
Now, when clicking on the Spectrum Xfer button and Odd>Even LED lights up, you will only hear the harmonic spectrum transfer from the odd bands applied to the even bands. And since the sine-oscillator does not contain any additional harmonics besides the fundamental, the pitch of the sine-oscillator now basically decides which of the even bands that is amplified. Try changing the pitch of the sine-oscillator and you'll see (and hear) the spectrum transfer follow.

Note that, while in Spectrum Xfer mode, morph or externally controlled CV is ignored (also described in further detail below).

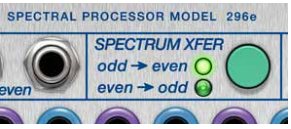
9. As a final lesson in this "getting started"-segment, let's look closer at the envelope follower outputs. Let's use a white noise source to feed the 296e again. To the right in this patch I'm setting up four sine oscillators with different tunings, each through their own VCA. The VCA themselves are driven from the envelope outputs of bands 4, 7, 10, and 13. The resulting sound is a kind of a "tuned particle noise". Now, by using the display envelope mode (as pictured) you can "draw" the decay curve differently for the different bands (band 4, 7, 10, and 13 used here).

Parameters

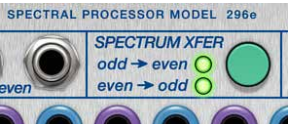
Spectrum Xfer mode button This button switches between the four different spectral transfer modes in the 296e. In normal mode (off), both of the Spectrum Xfer mode LEDs (odd>even and even>odd) are turned off meaning that no internal coupling is being made between the odd and even bands.



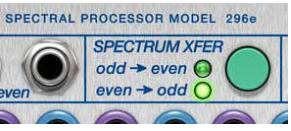
Clicking on the Xfer mode button once will light the odd>even LED and will internally couple all odd bands envelope outputs with all even bands CV inputs. In this mode, only the Even output will output audio.



Clicking on the Spectrum Xfer mode button again and both the odd>even and even>odd LED will lit up, indicating internal cross-modulation between all odd and even bands.



Clicking on the Spectrum Xfer mode button yet again will light up only even>odd LED indicating a odd to even internal coupling. In this mode, only the Odd output will output audio.



Clicking on Spectrum Xfer a fourth time, will turn off the spectral transfer altogether and the 296e is back in normal operation mode again.

Note: In all of the active spectral transfer modes, all external signals connected to the CV inputs are neglected, morph is not active and the envelope output signals will reflect the bands internal routing.

Display mode button Click this button to switch between the different display modes for the touch-display. Some modes are for input of data ("A Levels", "B Levels" and "Decay") while some modes only will show the results ("Progrm" and "Envelps"):

A levels In this mode, the touch-display is used for programing the odd and even levels for the Morph A. Morph can then be used to fade between the levels set in the A and B levels display pages (for further description see below).

B levels In this mode, the touch-display is used for programing the odd and even levels for the Morph B (for further description of Morph see below).

Progrm This mode uses the touch-display merely as a meter display for showing the resulting current output for the bands.

This means that the Progrm page in "normal mode" (no spectrum xfer mode activated) will show the levels applied to the VCAs for each band. This can be the blend of the A and B levels via the morph knob, as well as added external modulation via each band's CV input.

In Spectrum Xfer mode, the Progrm page will show the levels of the VCAs, but in this mode they're coming from the envelopes that are internally coupled while all other modulation (Morph blend, CV inputs) are neglected. See also further comment above in the Spectrum Xfer mode description.

Decay This mode uses the touch-display to set the envelope-follower decay time for each band. The response is exponential which means that the first bottom five LEDs on the touch display will have a very subtle change in decay, where the top three LEDs will have quite drastic changes. This is intentional and works the same way as the original hardware. Maximum decay time around 40 seconds.

Envlps This mode uses the touch-display to display the envelope-follower outputs for each band.

Note: You can also go between the different display modes by clicking on each mode's text. In this way it is possible to for example jump directly from A levels to Decay and vice versa.

MIDI enable This button enables MIDI mode. When MIDI mode is enabled, the 296e first listens to the incoming MIDI (blinking LED), locking on to any channel where a MIDI note on message is found (LED becomes solid). MIDI notes C0-D#1 plays the 16 bands with MIDI velocity controlling the amplitude of each band. There's also other repeating keyranges control at C2-D#3, C4-D#5, C6-D#7.

Note that in MIDI mode applied Morph is disabled. Also note that Spectrum Xfer modes will override MIDI mode input, meaning that MIDI note information will be ignored in Spectrum Xfer mode.

Morph Knob A-B This knob morphs between the A Levels and B Levels. The resulting output levels are displayed in Progrm mode (see description above).

Sig in Modulation This knob sets the amount of modulation applied from the signal at the "sign in" input.

Pre-Emphasis Odd This sets the amount of high frequency pre-emphasis applied to the Odd signal.

Gain Odd This is the gain amount of the Odd signal. It determines the loudness level of the signal input at the Odd input. Note that although the signal input at the Odd input is pre-patched to the Even input, the Gain knobs are not and thus still need to be set separately.

Gain Even This is the gain amount of the Even signal. It determines the loudness level of the signal input at the Even input.

Pre-Emphasis Even This set the amount of high frequency pre-emphasis applied to the Even signal.

Freeze This button is kind of special. It freezes the current state of all of the 296e levels applied to each band's output VCAs until clicked again; a toggle button functionality. This behavior is equal to holding down the freeze button on a real world hardware 296e. In addition to this, when toggling from freeze mode to normal mode in Level A, B, or Decay display mode, the spectrum of the current input signals is sampled to the memory of that particular page. Toggle button active is indicated by blinking LEDs on the touch display.

Note: When the 296e is in freeze state via the Freeze toggle button, no other buttons or touch-display entry will be accepted until freeze toggle button is turned off again (LEDs stop blinking). However, this does not apply to externally modulated freeze states (via the Pulse in jack) where the only parameter temporarily off limits to the user, is editing the touch-display while the 296e is sampling the audio spectrum into memory (during Pulse in jack at high, +5V).

Indicators

Spectrum Xfer mode LEDs When any of the Spectrum transfer ("Xfer") mode LED is lit, this indicates that envelope outputs of the Odd bands are coupled (pre-patched) to the even bands and vise versa. When both "Odd>Even" and "Even>Odd" LEDs are lit at the same time, this indicates full cross-modulation (not easily possible via external cables). When both Spectrum Xfer mode LEDs are turned off, "normal" mode (no pre-patching) is activated.

Display mode LEDs Indicates current state of the touch-display (A levels, B levels, Progrm, Decay, or Envlps).

MIDI mode LED This LED indicates MIDI mode activated (LED is lit), MIDI learn (LED is blinking while the 296e is waiting for a MIDI note on message on a MIDI channel) or MIDI mode off (LED is unlit).

Program Level1-16 A levels, B levels – these are two differently programmed sets of output levels of the 296e that can be setup to be morphed between. Morph means that the two sets of output levels are continuously changed into the other at the turn of the Morph knob or by applying external modulation to the Morph CV input. The resulting (Morph) curve is displayed in Program mode ("progrm" selected). When Decay mode is selected, the touch-display will show the envelope decay times for each band and in Envelope mode ("envlps" selected), the resulting output envelopes CV outputs for each band is displayed. If the Freeze function is manually latched by the toggle button, frozen bands will be indicated by blinking LEDs on the touch display.

Envelope outputs 1-16 Envelope display mode (Envlps) shows the output state of the envelope-followers at each band on the touch-display (but with no touch in this mode as it is purely for visual feedback).

Morph LEDs The Morph LEDs indicate the transition between the A levels and B levels, from Morph A LED fully lit while Morph B LED is unlit to gradually the opposite.

Inputs

CV input 1-16 These are the CV inputs for controlling the attenuated output of 296e's 16 bands. The CV signal applied here is summed to that of the drawn A and B level curves.

Morph CV in This is the CV input for external control over the 296e Morph functionality. Any CV input here is added to the offset set by the knob.

Audio Mod in (sig in) This is the audio input for external modulation of the 296e Morph function. Any signal input here will do.

Odd Signal in This is one of the two audio inputs for the 296e, the input for the odd bands. Both of the inputs are normalized to each other which means that if no cable is inserted in the other input jack, a signal input to either one will be fed to both. For example, if a signal is inserted at the Odd Signal in jack without any cable present at the Even signal in jack, the same signal will be fed to all bands - odd and even - of the 296e. Note that both sides still have their own dedicated gain and pre-emphasis gain controls though.

Even Signal in This is the other of the two audio inputs for the 296e, the input for the even bands. The same description as for the Odd Signal in jack applies here (see above).

Pulse in This CV input is the remote control of the Freeze functionality. As soon as a logical high (CV above +2.5V) appears on this input, the Freeze function is engaged which is indicated by flashing LED display levels.

Outputs

Odd Comb out This is the comb filter outputs for all odd bands before internal attenuation.

Even Comb out This is the comb filter outputs for all even bands before internal attenuation.

Odd out This is the attenuated output for all odd bands.

All out This is the attenuated output for all bands, odd and even. This is to be considered the most commonly used output.

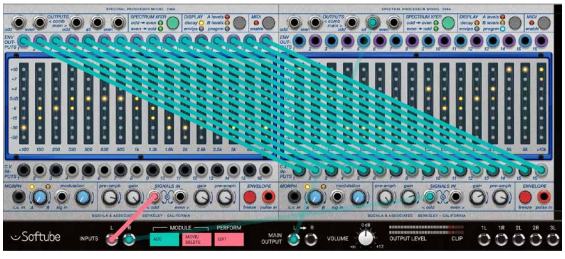
Even out This is the attenuated output for all even bands.

Envelope out bands 1-16 These are the CV outputs from each band's envelope follower. These levels are affected by inputs signals, internal or external linking or attenuation and the decay times set in the Decay display mode.

Note: The visualisation of the CV outputs at each Envelope out band is available in Envelope display mode ("envlps" selected).

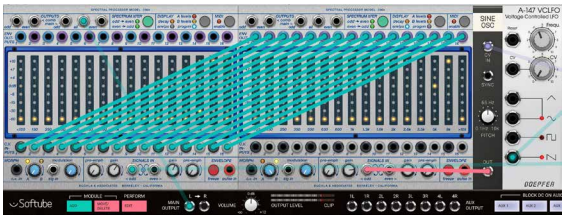
The Buchla 296e module in use

In this patch we have connected a rhythmic source like a drum loop to the left inputs of Modular FX and a pad-like loop to the right inputs. By using the Spectrum Xfer mode odd>even and only listening to the even outputs, a crude version of a vocoder is created (8 band vocoder). Try adjusting the Decay times at the Decay page for different flavors.

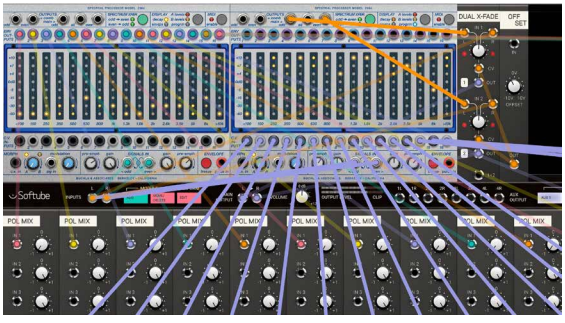


We also use multiple instances of the buchla 296e together to create a "fuller" vocoder sound than described above. The left 296e is fed by the rhythmic part coming from the left input and the right 296e is processing the audio coming in through the right input. The key thing here is the envelope outputs of the left 296e that are routed to the CV inputs of the bands of right 296e.

The 296e's envelopes can of course be used to extract beat information out of an audio-track fed into Modular FX. Here's an example of how you could set up the 296e together with some Heartbeat-modules to produce a drum-replacer.



In the original 296 from the Buchla 200-system there was a function called programmed control that let you use the 296 as a bandpass focusing filter. The patch pictured above is something in that direction with an analyser part (right) that tracks the sineocillator and a carrier part that filters the A-110 VCO.



This is an interesting concept - using one 296e as analyser (left) and then using the inverted envelope signal for each band to reduce gain off the second 296e with all levels at full. This gives us control over the ducking decay for each band as well as a parametric EQ at the same time. The Dual X-fade and offset setup at the right is for blending some original signal back in.

You can also take advantage of the 296e's analysing capabilities and use the envelope as a kind of "audio-driven sequencer". In this patch we use the pitch of the top left sine oscillator to drive the 296e, but of course any type of audio could be used.

Credits

Jacopo Lovatello – programming, modeling. **Kristofer Ulfves** – product owner, testing, preset, user manual. **Kim Larsson** – mentoring. **Björn Rödseth** – mentoring. **Arvid Rosén** – mentoring. **Bitplant** – GUI graphics. **Fanny Hökars** – user manual layout.

13 Chandler Limited® Curve Bender

CHANDLER LIMITED'S FOUNDER AND CHIEF designer **Wade Goeke** had many ideas for reimagining the historic console EQ to work in a modern context. After experimenting with a number of prototypes, Goeke incorporated several new features into the equalizer, creating what is now considered a modern classic — the **Chandler Limited Curve Bender EQ**.

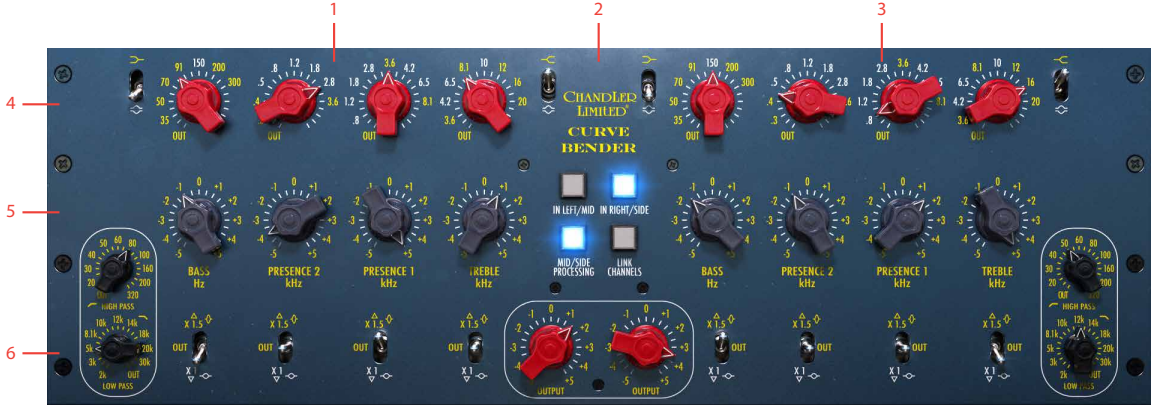
Vintage Sound, Modern Features

That beast was called the **Chandler Limited EMI TG12345 Curve Bender**, and it was issued in celebration of the 75th birthday of Abbey Road Studios. Powerful functionality gives you serious sound shaping mojo. Wade Goeke turned the nine fixed frequency selections of the original Abbey Road desk into 51 EQ points. He also added the **Multiply** switch which increases the boost/cut ability from 5 to 15 dB, as well as sharpening the Q. He furthermore added high and low pass filters, and bell/shelf selections on the high and low bands.

So the Chandler Limited EMI TG12345 Curve Bender® is a classic equalizer reimagined from the glorious history of EMI and Abbey Road Studios which has come full circle, morphing into a modern powerhouse EQ perfect for all your recording, mixing, and mastering needs.

> The Curve Bender's additions to the original TG12345 feature set are marked with yellow text color on the panel. The original features have white text color, except the Mid/Side and Link features.

User interface



1. Left/Mid section
2. Center section
3. Right/Side section
4. Top row
5. Middle row
6. Bottom row

Parameters

Top row, left to right (duplicated for the RIGHT channel)

Bass Shelf/Bell switch Switches the **Bass** band between SHELF or BELL operation.

Bass Frequency Selects the operating frequency of the **Bass** band. The available frequencies are 35, 50, 70, 91, 150, 200 and 300 Hz. The band is bypassed in the OUT position.

Presence 2 Frequency Selects the operating frequency of the **Presence 2** band. The available frequencies are 0.3, 0.4, 0.5, 0.8, 1.2, 1.8, 2.8 and 3.6 kHz. The band is bypassed in the OUT position.

Presence 1 Frequency Selects the operating frequency of the **Presence 1** band. The available frequencies are 0.8, 1.2, 1.8, 2.8, 3.6, 4.2, 6.5 and 8.1 kHz. The band is bypassed in the OUT position.

Treble Frequency Selects the operating frequency of the **Presence 2** band. The available frequencies are 3.6, 4.2, 6.5, 8.1, 10, 12, 16, and 20 kHz. The band is bypassed in the Out position.

Treble Shelf/Bell Switch Switches the Treble band between SHELF or BELL operation.

Middle row, left to right (duplicated for the RIGHT channel):

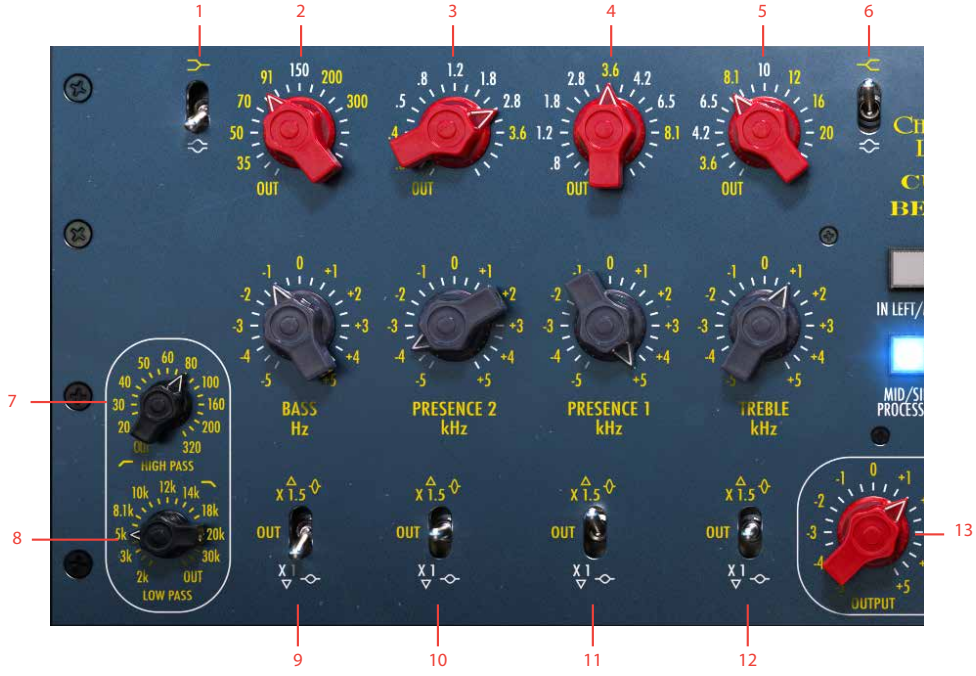
High Pass Selects the operating frequency of the **High Pass** filter, which has a fixed slope of 6 dB per octave. The available frequencies are 20, 30, 40, 50, 60, 80, 100, 160, 200 and 320 Hz. The filter is bypassed in the Out position.

Low Pass Selects the operating frequency of the **Low Pass** filter, which has a fixed slope of 6 dB per octave. The available frequencies are 30, 20, 18, 14, 12, 10, 8.1, 5, 3 and 2 kHz. The filter is bypassed in the Out position.

Cut/Boost Cuts or boosts the selected frequency with a range of either +/- 5 dB or +/- 15 dB, depending on the setting of the corresponding **Multiply** switch.

Bottom row, left to right (duplicated for the RIGHT channel):

Multiply switches When set in the bottom position, cutting and boosting will have a maximum range of 5 dB. In the top position, the boost/cut range is increased to 15 dB to enable more aggressive equalization. This also decreases the width (Q) of the EQ curve for tonal variety. With the **Multiply** switch in its center position, the corresponding band is bypassed.



1. Bass shelf/bell
2. Bass band
3. Presence 2 band
4. Presence 1 band
5. Treble band
6. Treble shelf/bell
7. High pass
8. Low pass
9. Bass: frequency, cut/boost, multiply
10. Presence 2: frequency, cut/boost, multiply
11. Presence 1: frequency, cut/boost, multiply
12. Treble: frequency, cut/boost, multiply
13. Output gain

Center Section

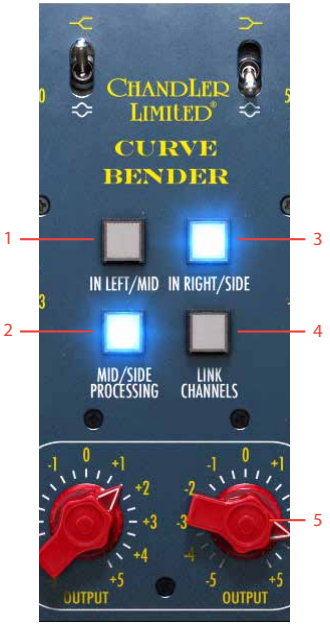
In Left/Mid Activates or bypasses Curve Bender's processing of the LEFT or MID channel (depending on the position of the **Mid/Side Processing** button) to facilitate direct comparison of the processed and unprocessed sound.

In Right/Side Activates or bypasses Curve Bender's processing of the RIGHT or SIDE channel (depending on the position of the **Mid/Side Processing** button) to facilitate direct comparison of the processed and unprocessed sound.

Mid/Side Processing Enables MID/SIDE processing. When in MID/SIDE mode, the LEFT channel of the equalizer controls the MID channel, and RIGHT channel controls the SIDE channel.

Link Channels Links the LEFT and RIGHT channels in STEREO mode, and MID and SIDE in MID/SIDE mode. When the channels are linked, any adjustments made to one channel will be automatically reflected in the other.

Center Section



1. In left/mid
2. Mid/side processing
3. In right/side
4. Link channels
5. Output gains

Output Gain Controls the output gain in 0.5 dB steps. Adjust this so the output level is reasonably matched to the unprocessed sound of the original sound source. This lets you accurately perform A/B comparisons of the Curve Bender bypassed and active.

In Use

The Curve Bender® alters the tonal content of the sound source in a musical, yet transparent, manner. It is equally useful during recording, mixing or mastering. The most common use is probably on a stereo source, with the unit in stereo mode and its left and right channels linked. This means that exactly the same processing is applied to both the left and right side of the sound source. But it works equally well on mono sources, and the ability to unlink the left and right channels means you can apply different equalization to the different sides, if necessary. For example, a stereo recording might have been made with slightly unmatched microphones—this can be remedied by unlinking the Curve Bender's channels and apply equalization to match one side to the other.

Mid/Side Tips

The Curve Bender® plug-in offers a mode which is not available on the hardware—the **Mid/Side** mode (MS).

In **Mid/Side** mode, the Curve Bender® will not process the left and right channels separately. Instead, the sound is separated into a MID portion of the sound, which contains only the material which is common to both left and right channels (the mono portion of the sound). The other portion is the SIDE portion of the sound, which contains only the material which differs between left and right.

Mid/Side for Mastering

Mid/Side processing is typically useful in a mastering situation, where you are working on stereo program material without access to the individual channels. For example, the lead vocal is typically placed firmly in the center of the stereo panorama and thus entirely resides in the MID portion of the sound, and not at all in the SIDE portion. So if you find a need to apply equalization to the lead vocal, but wish to avoid affecting the other instruments in the same frequency range, you can set Curve Bender® to MID/SIDE mode, and only adjust the MID portion to achieve the desired effect on the vocal. Doing it this way leaves the SIDE portion untouched—where you in many cases will find panned guitars or stereo keyboards. You can of course also do the opposite, only affect the SIDE portion of the sound.

Stereo Widening

Mid/Side processing can also be used to make a stereo sound appear wider. You do this by slightly turning down the MID portion of the sound—but don't go too far as you might end up with a center "hole" in the mix.

Mono/Stereo Operation

The Curve Bender® is a stereo unit, but can also be used in MONO or MULTI-MONO.

Please note, when using Curve Bender® in MONO or MULTI-MONO, make sure the channels are linked and **Mid/Side Processing** is disengaged.

Credits

Kim Larsson, Arvid Rosén - modelling. **Niklas Odelholm** - graphic design. **Paul Shyrinskykh** - product owner, manual. **Patrik Holmström** - framework programing. **Ulf Ekelöf** - 3D rendering.

ABBEY ROAD STUDIOS, EMI, TG, CURVE BENDER AND THEIR ASSOCIATED LOGOS ARE TRADEMARKS OF EMI (IP) LIMITED.

14 Chandler Limited® Germanium Compressor

"… like all the Germanium units, many of the parameters have been set by extensive listening and recording, not by technical analysis and equations. We feel this approach leads to very musical equipment that is more similar to a fine instrument. In many cases we actually TRIED NOT to "accidentally" discover the physics and math behind what we were doing, so that that nothing got in the way of what we were hearing :-)

This compressor has been in the design process for over two years. We really hope you'll like it!!!"

Wade Goeke, founder, owner and chief product designer at Chandler Limited®

The Germanium Compressor is like no other compressor. It is really different. The way it grabs the signal, the way it colors it and the way the knobs respond when you tweak them is different from other compressors we come across, and we've come across quite a few. It was also extremely hard to model, and that's largely because of the method with which Wade Goeke at Chandler Limiter® designed it. Instead of looking at hardware specs, noise floor, control theory and the math, Wade Goeke designed this almost exclusively by using extensive listening tests. The resulting circuits are therefor not always easy to understand, although the character of this compressor is obvious and in your face.

But the real master piece here is not just to design something different with lots of mojo, but to design something that's also extremely versatile. And Chandler Limited® managed to do that.

This is a compressor that excels at drums, vocals, bass, etc. but also at mix busses and masters. We hope you will enjoy it, because we are very proud of being able to bring this to you in software form.

Quoted parts in the manual are taken directly from the hardware manual of the Chandler Limited® Germanium Compressor

User interface

1. Left/Mid section
2. Right/Side section
3. Signal routing

Parameters

The user interface consists of two identical units, the top one for the left channel and the bottom one for the right. If you always want the same settings on both channels, just turn on **Link Channels** and the settings will be the same for both channels.

The lowest part of the user interface has an extra panel for routing the signal (**Mid/Side Mode, Link Channels**), external sidechain as well as an output volume control.

Compressor Parameters

Clean Comp/ Dirty Comp The gain reduction element in the Germanium Compressor consists of a FET transistor, and these are known for the high amount of distortion. The design of the Germanium Compressor compensates for this, but if you want the dirty and uncompensated sound of the FET gain reduction, set this switch to DIRTY COMP

Input Sets the amount of signal into the compressor, turn it up and get more compression. Depending on the gain staging you use in your DAW you might need more or less input gain to achieve the desired amount of compression.

In the software version of Germanium Compressor we have increased the range of the **Input** knob so that it's easier to use.

Sidechain Sidechain filter setting. Use this to set how much bass the compressor should react to. Frequencies below the set frequency is attenuated and will trigger the compressor less. When ever you use the compressor on something with a wide frequency range, such as a drum bus, a mix or sub groups, experiment with the setting of this control. It's almost magic!

Ratio Controls the ratio of the compressor. Keep in mind that the original hardware was designed extensively through listening tests, as opposed to a mathematical approach. Instead they adjusted the individual settings while listening to various pre-recorded tracks and used what sounded best on most sources.

The **Ratio** works intimately with the Comp Curve control. It is recommended that you first select the sound of the **Comp Curve** (the knee) and use the ratio to fine tune the compression from there.

Comp Curve One of the most fun and most original part of the Germanium Compressor. The curve adjusts the knee of the compression with various diode combinations.

"Designing this section was one of the hardest and most rewarding parts of the Germanium Compressor. Simply put, we took every type of diode we could find and endlessly experimented with them in the curve section until we found the most useful combinations. Most Ratio/Curve controls are simple resistance circuits. Using diods add flavor and character. We ended up with six different combinations, using germanium, silicon and zener diodes in several configurations as well as a more standard resistance setting.

The individual choices were selected to give the best range of sounds on a variety of sources. Germanium diodes have a low forward voltage which translates to a very smooth compression knee. Silicon forward voltage is higher and makes the knee sharper. Zener diodes have another voltage characteristic that has an even more extreme effect on the knee."

R SOFT: The most neutral and gentle of the curves. It uses a standard resistance circuit and has the least coloration.

GERM SOFT: Uses a single germanium diode, that has a gentle diode curve. Slightly sharper knee than R SOFT, but with a nice color and vibe.

GERM MED: Five germanium diodes of another type than the one used in GERM SOFT. Even sharper knee, and a bit more vibe.

SILICON MED: A single silicon diode, has a sharper knee than GERM MEDIUM, but a different tone.

SILICON HARD: Two silicon diodes, an even sharper knee and more vibe.

ZENER HARD: Uses a single Zener diode with the sharpest knee of all settings. It also has a very different character compared to the other settings.

"Simply put, the knee of the curve gets sharper as you turn it up. Each diode has its own flavor as well. The R SOFT, GERM SOFT, GERM MED and SILICON MED tend to be the best for general use with each having its own tonality. For compression with the least artifacts or audibility choose the R SOFT or GERM SOFT settings. For a little more coloration and character, choose the GERM MED and SILICON MED.

GERM SOFT and GERM MED selections tend to be our favorite general use settings as they are the best compromise between coloration, tone, and punch. On drums, for example, you can use larger amounts of compression without losing the attack, punch and tone of the drum, while the germanium diode adds character.

For stereo buss or mastering, use the R SOFT or GERM SOFT settings. For hard smash, use the SILICON HARD or ZENER selections. For submixing with the **Mix** control, use the harder presets."

Mix Blend between the compressed (WET) and un-compressed (DRY) signal. Mix the tight compressed sound together with the unprocessed original signal and get the best of both.

The dry signal is still affected by the tone of the **Germanium Drive** output circuit, so if you want to play around with that without any compression, just set **Mix** fully to DRY.

"A couple of favorite uses… For intense sub-mix sounds, use the HARD SILICON and ZENER curves with a faster attack. Set the **Mix** control on WET from three to zero.

For a more subtle sub-mix sound, where you just want to keep the track from sounding too soft, use the moderate curves with a medium attack. Mix in just enough dry signal to add some life back to the compressed sound.

One excellent use of this is to use two compressors in series. The first would be the Germanium Compressor set with the **Mix** at slightly DRY. You then use the second compressor to compress the remaining unprocessed sound. Interesting control of the tone as well as the dynamics can be achieved."

Attack Sets the attack time of the compressor. Goes from "relatively fast" to "very slow".

Release Sets the release time of the compressor.

"**Attack** was designed by listening to as much material as possible. We have never calculated the actual attack and release times in an attempt to keep them as musical and natural as possible. Another important feature of the **Attack** is the range it has. It is relatively fast, but more importantly it can be very slow! We found this extremely useful on drums, bass, guitars, mixes and mastering. For example on drums you can control the attitude, punch, and intensity by the attack setting. For tight drums, setting the attack from medium to slow lets you compress large amounts which will make them very punchy and will not overly soften the sound by dampening the attack. The same results apply with bass, etc. For general use we found medium attack times (2-6) very appropriate."

Germanium Drive A gain switch in 3 dB steps. This is an output level control with built-in character. The actual amount of overall gain is affected by both the **Germanium Drive** and **Feedback** controls.

Using **Germanium Drive** and **Feedback** in different combinations is essential if you want to take advantage of all the different tones of the Germanium Compressor.

Feedback Sets the vibe and tone of the output stage. At 0 you have more high frequencies, less distortion, less gain and a clearer tone. Increase it to get more distortion, gain, and a great deal of character in the bass and treble ranges.

On the original hardware, the effect of the **Feedback** control depends heavily on the equipment the Germanium Compressor is connected to, more precisely the input impedance of that unit. In the software version of the Germanium Compressor, we chose to "connect" it to an industry standard +4 line input, which is what a large majority of Germanium Compressor users do.

Routing Parameters

These parameters control the routing of the compressors, and allow you to link/unlink the compression, route mid/side, or use an external sidechain.

External Sidechain External sidechain is engaged when this switch is in its up-most position.

Ext S/C Gain Sets the gain of the input on the external sidechain.

In Left/Mid
In Right/Side Use to bypass the compressor.

Mid/Side Processing Switches between Left/Right (normal) processing and Mid/Side processing.

In Mid/Side processing the top compressor is processing the Mid part of the input, which contains the mono information. The lower compressor is processing the side information. Mid/Side is extremely useful when working on drums buses or mixes. A small room can sound huge when processed in Mid/Side.

Please note that you need to unlink the channels (**Link Channels** set to OFF) to get true Mid/Side operation.

Tip: On a drum bus, turn on Mid/Side processing and dial-in a punchy 3-5 dB compression on the mid part. On the sides, use fast attack and release times to squash the signal as much as you can, while dialing in some dry using the Mix control. Use the Germanium Drive to balance the room tone (the side compressor) with the punch (the mid compressor).

Link Channels Links the controls and the sidechain of both compressors.

Linked controls are just for your convenience, but the linked sidechain has a profound effect on the sound. With a linked sidechain, both sides will always get the same amount of gain reduction, and a loud sound in one channel will affect the gain of both. This is the preferred method when using much compression, because the sound stage might tilt to the left or right otherwise.

When using less compression, or mastering applications, it might be interesting to de-couple the left and right sides and get a more pronounced stereo image and "depth" that way.

Output Volume Sets the output volume on both units. A completely colorless volume control from -24 to +24 dB.

Linked vs. Unlinked Channels

Normally, a stereo compressor will have the same amount of gain reduction on both channels, regardless of the difference of the channels. This is to, for example, avoid having the entire sound field tilt to the left when a loud sound is heard in the right channel. But for mastering purposes, and lower gain reduction amounts, it is often very useful to decouple these two channels in order to preserve a "depth" to the sound.

By engaging **Link Channels**, you'll get the "normal" stereo mode with linked gain reduction, as well as linked parameters. Set **Link Channels** to off and you get true, independent gain reduction.

This feature is even more important in **Mid/Side Processing** mode, where you seldom want to link the gain reduction.

Linking Best Practices

STEREO MODE: If you have lots of gain reduction, link your channels.

MID/SIDE MODE: Unlink them.

STEREO MODE: If you are mastering or applying small amounts of gain reduction, try to unlink them. But look out for sudden shifts in the stereo image related to the gain reduction.

As always, trust your ears. If it sounds good, it is good!

Credits

Hardware Modeling: **Jacopo Lovatello** (lead), **Björn Rödseth, Kim Larsson,** and **Arvid Rosén**

Original Hardware Design: **Wade Goeke**

GUI Design: **Niklas Odelholm, Ulf Ekelöf**

A/B and QA Testing: **Johan Bremin, Jacopo Lovatello, Wade Goeke,** and **Niklas Odelholm**

Manual: **Niklas Odelholm**, with excerpts from the hardware manual by **Wade Goeke**

Project Management: **Niklas Odelholm, Paul Shyrinskykh**

15 Console 1

Softube Console 1 is a new type of audio mixer. It is a hardware used together with your favorite digital audio workstation (DAW), and just like a traditional mixing console, you can route any or all DAW tracks to Console 1 and mix them from within that system. This gives you access to Console 1's world class sounding effects sections and lets you control everything from the hardware unit, using physical knobs and buttons.

There is a separate manual for Console 1. You'll find it by clicking on the Console 1 menu item, or on www.softube.com.

So what Console 1 gives you is the great sound that has made Softube world famous, in addition to a tactile and intuitive workflow that speeds up your mix work substantially compared to mouse mixing on the computer.

The SSL SL 4000 E Channel

Included in your purchase is Softube's model of a Solid State Logic SL 4000 E, one of the most famous and coveted mixing consoles in the world. SL 4000 E consoles have been installed in hundreds of the world's leading studios, and it is said that Solid State Logic's 4000 series have been used on more platinum selling records than all other console models combined! This is for good reason. The SL 4000 E has a signature sound that is transparent enough to work for all music styles, but still adds its touch of luxury. Things simply sound like a record when they've been run through the SL 4000 E.

With Console 1, you get the famous black knob E242 equalizer, the classic channel compressor, the very musical gate/expander and the sweet high/low cut filters from the SL 4000 E channel. We also modeled the unique harmonics, distortion and non-linearities you get from running the SL 4000 E into overdrive. Controlled by Console 1's Drive knob, you can use it subtly on all channels to glue the mix together like a slightly overdriven SL 4000 E does, or not so subtly to add edge and presence to selected sounds.

In addition, you get Softube's unique Transient Shaper, which is not modeled off the Solid State Logic SL 4000 E or any other unit for that matter.

Softube's model of the Solid State Logic SL 4000 E channel is only available as part of the Console 1 system and not as a single plug-in. Softube will continuously release more channel strip models to be used with Console 1, that can be purchased separately.

More Information

For more information, please visit `www.softube.com` or the Console 1 specific manual.

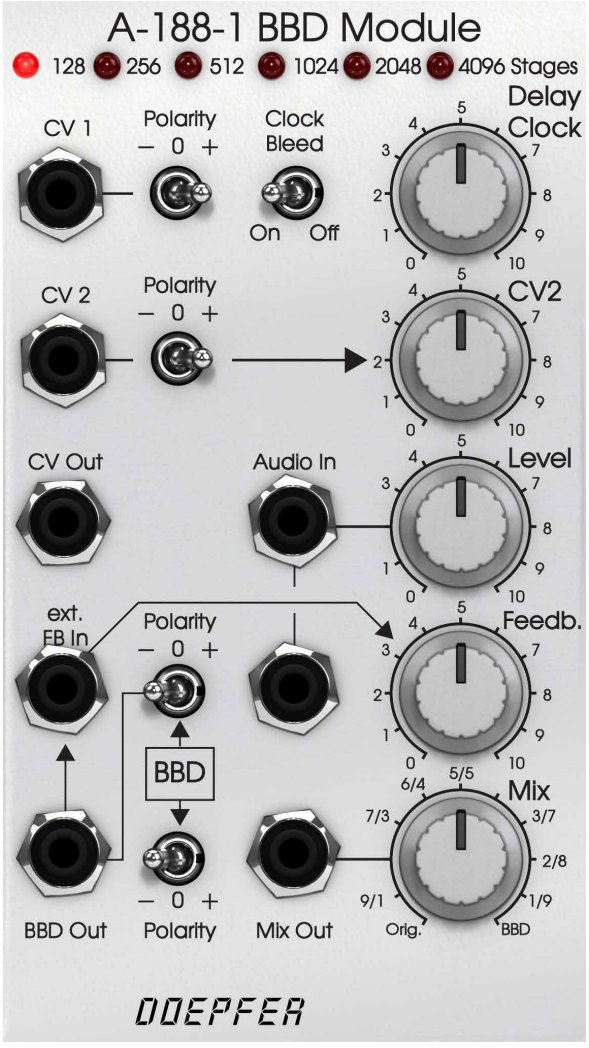
MODULAR EXCLUSIVE

16 Doepfer A-188-1 BBD module

The Doepfer A-188-1 BBD module for Softube Modular is an emulation of a so called Bucket Brigade Device, a kind of early "digital" memory used in the 70s and 80s in all kinds of delay-based effects such as choruses, flangers, ensemble-effects and of course "analog" echo-units.

A Bucket Brigade Device can be described as a long row of level-memories that pass their position at a high speed providing a rudimentary form of digital memory. The speed at which these memories are passed through the module is determined by an internal clock, normally run at very high speed. The length of the memory held is then determined by the number of stages in the BBD-chip used in the module. The BBD chips have a very specific working area that only guarantees correct behavior (no degradation of the sound) above a certain clock-frequency. This is why gradual degradation of the delayed signal occurs when the clocking-frequency of a BBD chip is lowered.

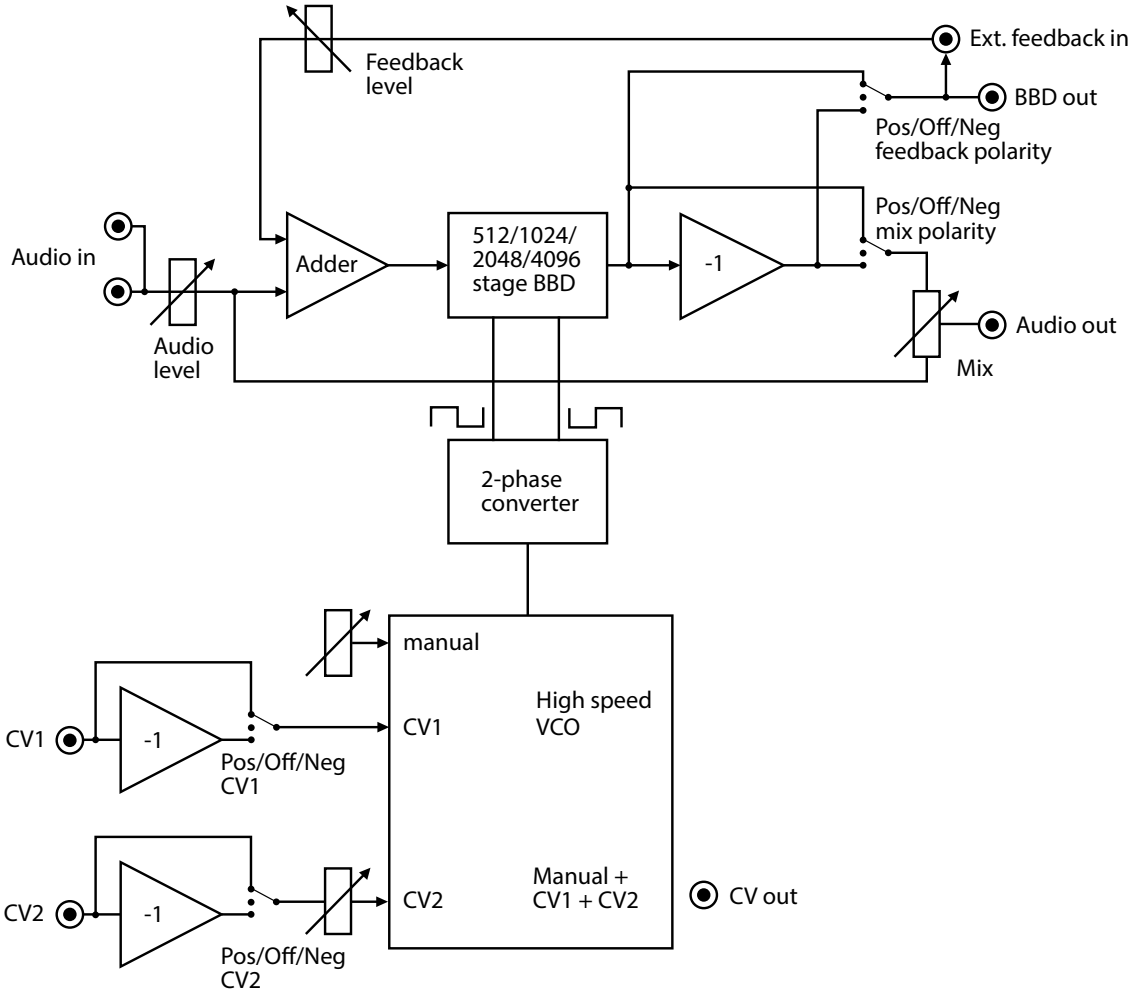
The Doepfer A-188-1 hardware module is normally shipped with one specific BBD chip in order to fit a specific purpose. Some modules ship with a chip with 128 stages, some with 256 stages and so on. The Doepfer A-188-1 hardware we have used came with 1024 stages using a modern BL 3207 BBD chip inside, and was thus the basis of the model used in this Softube emulation. This chip is a bit noisy and the module has apparent clock-bleed through when lowering the clock-frequency, so we have therefore made this noise and clock-bleed optional via the new Clock-bleed switch. Another change from the hardware version is the absence of the clock out and external clock input, simply because it was not possible to emulate and support the extremely high clock-frequencies in the MHz area within Modular.



Overview

The Doepfer A-188-1 BBD module has been designed with experimentation in mind. This is why the signal going through the BBD chip is not filtered or processed in any way (as is often the case in dedicated units such as choruses). Thus the output signal is deliberately rough sounding and full of aliasing artifacts when input is pushed too high or internal clock-frequency is lowered. At full feedback the module can self-oscillate or "scream" in a distinctive (pleasing) manner.

The internal clock can be CV-controlled via the CV1 or CV2 input jacks (the latter also scaled via the CV2 attenuator knob). Built into the A-188-1 BBD is also a feedback path with positive or negative feedback controlled via the feedback attenuator knob. This path can be broken and fed via external modules in order to gain control of this aspect as well (see further examples below).


Feedback level
Ext. feedback in
BBD out
Pos/Off/Neg feedback polarity
Pos/Off/Neg mix polarity
Audio in
Adder
512/1024/ 2048/4096 stage BBD
-1
Audio level
Audio out
Mix
2-phase converter
manual
High speed VCO
CV1
-1
Pos/Off/Neg CV1
CV2
-1
Pos/Off/Neg CV2
Manual + CV1 + CV2
CV out

Parameters

Stages buttons 128/256/512/1024/2048/ 4096: these buttons determine how many stages of BBD chip is emulated with the A-188-1 BBD module. Fewer stages tends to work well for short delay-based effects such as flangers and choruses, while more stages work better for longer echoes and reverb-like effects.

Polarity switch CV1 This switch sets the effect incoming control voltage in the CV1 input will have on the internal delay clock speed. When the polarity switch is set to "-" incoming CV signal through the CV1 jack will be inverted, while when polarity switch is at "+" CV is positive. "0" position turns incoming CV off on the CV1 input.

Polarity switch CV1 This switch sets the effect incoming control voltage in the CV1 input will have on the internal delay clock speed. When the polarity switch is set to "-" incoming CV signal through the CV1 jack will be inverted, while when polarity switch is at "+" CV is positive. "0" position will turns incoming CV off on the CV1 input.

Clock Bleed switch This switch turns on and off the authentic clock-bleed and noise that is part of the original Doepfer A-188-1 hardware. We thought it would be nice to have the possibility to get rid of these sound artifacts so we added this switch in the place where the original hardware has its Clock out and external clock input jacks (see above).

Delay Clock knob This knob sets the internal clock speed for the Doepfer A-188-1 and thus also the amount of delay introduced as the signal is stored and carried through the bucket brigade stages. This knob set at high values or fully clock-wise will cause short delays while preserving the fidelity of the original signal. With lower values, longer delays will be produced but also less fidelity in the resulting delayed sound. You will also notice hissing noise and apparent clock-bleed when lowering the delay knob while the clock-bleed switch is in the "on" position.

Polarity switch CV2 This switch works similar to the Polarity switch for CV1: It sets the effect incoming control voltage in the CV2 input will have on the internal delay clock speed. When the polarity switch is set to "-" incoming CV signal through the CV2 jack will be inverted, while when polarity switch is at "+" CV is positive. "0" position turns incoming CV off on the CV2 input. Notice that the control voltage input through CV2 input is scaled via the CV2 attenuator knob described below.

CV2 knob The CV2 attenuator knob scales the incoming control voltage on the CV2 input jack. When turned fully clockwise, the full input control signal is affecting the delay clock speed.

Audio input Level knob This knob set the level of the audio input. This knob can also be used to increase the audio level beyond the point where the BBD starts to distort which might be desirable in some cases.

Feedback Polarity switch The Feedback Polarity switch will determine whether the wet BBD signal fed back to the input is polarity flipped or not. Negative ("-") value will result in a different sounding feedback than a Positive ("+") one. When the Feedback Polarity switch is set to "0", no feedback is active in the module.

Feedback knob This is the attenuation of the wet BBD signal sent through the feedback path. When set at fully clockwise, attenuation is 0% which means that feedback will go well into self-oscillation.

Mix out Polarity switch The Mix out Polarity switch will determine whether the wet BBD signal sent to the Mix Out jack is polarity flipped or not. Negative ("-") value will result in a different sounding feedback than a Positive ("+") one. When the Mix out Polarity switch is set to "0", no wet BBD component will be heard in the mix output.

Mix knob This knob sets the mix between the clean signal and the wet, delayed, signal for the Mix Out jack.

Indicators

Stages buttons indicators The stages LED buttons light up to indicate which number of stages is selected.

Inputs

CV1 This is the first control voltage input for modulating the delay time from an external modulation source.

CV2 This is the second control voltage input for modulating the delay time from an external modulation source.

Audio in These are the dual audio inputs (one to the left of the Audio input Level knob and one to the left of the Feedback knob). Inputting audio into both will create a 50-50 mix of the two inputs before the Audio Level attenuator knob.

Ext FB in External Feedback input. This is the input for an external feedback loop where you for example could insert a VCA to CV control feedback (see user example below).

Outputs

CV out This is the tracking CV output for the delay clock frequency. This is great for example to create external anti-aliasing filters with CV controlled cutoff that track the delay time.

BBD out This is the fully wet (effect) output of the BBD module. Great as the output entry point for an external feedback loop.

Mix out This is the main output for the mixed signal between the dry (input) and wet signal of the Doepfer A-188-1 BBD module.

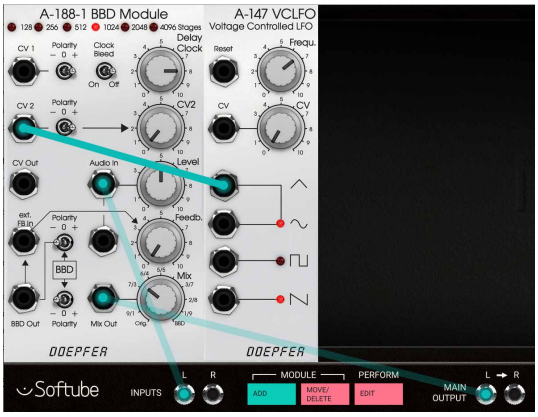
The Doepfer A-188-1 BBD module in use

The Doepfer A-188-1 BBD can be used for creating all kinds of time-based effects. Everything from obvious echos and reverb-like effects to choruses, flangers and ensemble effects. But the Doepfer A-188-1 can also be used as crude distortion effect or feedback oscillator for Karplus-Strong synthesis.

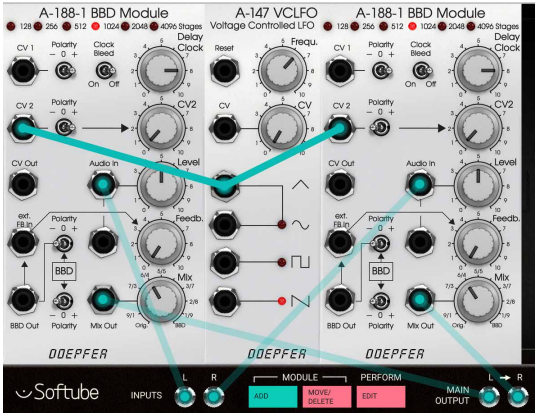
The most obvious effect the Doepfer A-188-1 BBD module is used for is creating a simple echo effect. This is done by using the 4096 mode and lowering the delay rate to a figure somewhere between 3 and 6. Remove the clock-bleed if you like by flipping off the clock-bleed switch or by external low pass filtering.

A pingpong echo can also be achieved by using two BBD modules and an external mixer for the mixing of the dry and delayed signal.

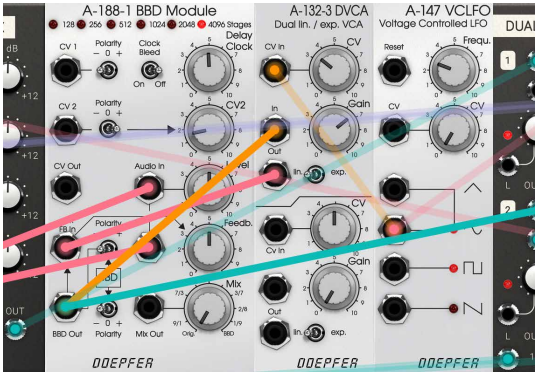
Building a simple chorus using the Doepfer A-188-1 BBD only requires an external LFO to create some movement in the short delay line.



A Stereo chorus is just an extension of the patch described above. Just add another BBD-module and feed that from the same LFO, but with inverse phase. This is done by flipping the CV2 Polarity switch to "-".



The BBD Out and ext. FB in jacks can be used to attach an external feedback loop to the BBD module. In this example an external Doepfer A-132-3 VCA module is used together with a Doepfer A-147 LFO to periodically control the feedback of the short echo created by the BBD module.

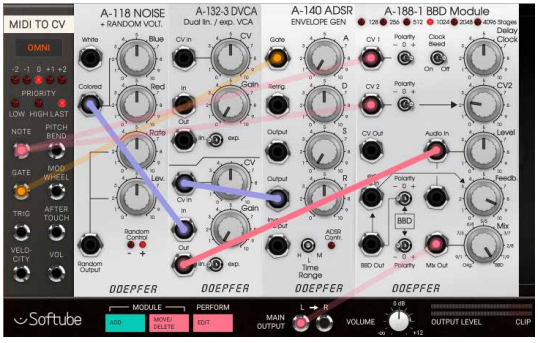


In this example we're using an external tracking low pass filter CV in the feedback path in order to remove clock bleed and get a "muddier" and "dubby" echo sound. Notice that the filter cutoff frequency now perfectly tracks the delay length, so that short echoes are brighter while longer echoes are duller with less high end.

Creating a flanger patch with the Doepfer A-188-1 BBD module is very easy. All we need are to use short delay times (using 128 stages here), and to sweep that delay time up and down from an LFO. Experiment with modulation depth by adjusting the CV2 knob and feedback by adjusting the feedback knob.

The Doepfer A-188-1 BBD module has a very distinctive distortion that can be utilised for striking lo-fi effects. Use very few stages (128 or 256) and change the delay length for into even more lo-fi, bit-crunchy territory. In this example we also use an external mixer to push the input volume even harder..

The Doepfer A-188-1 BBD module can be used for Karplus-Strong synthesis. Pictured here is a patch where a short burst of noise is used to trigger the BBD module acting as a resonating string. The CV inputs are adjusted to enable this patch to track somewhere around an octave, adjustment for broader ranges could be tricky though.



Credits

Jacopo Lovatello – programming, modeling. **Arvid Rosén** – modelling, mentoring. **Kristofer Ulfves** - product owner, testing, preset, user manual. **Niklas Odelholm** – GUI graphics. **Fanny Hökars** – user manual layout. **Igor Miná** - hardware photos.

MODULAR EXCLUSIVE

17 Doepfer A-101-2 Vactrol LPG

THE DOEPFER A-101-2 VACTROL LPG IS a vactrol based combination of a 12 dB Low Pass filter (LP) and VCA. This combination is often called *Low Pass Gate* (LPG), a term coined by synth legend **Don Buchla** for the **Buchla 292** module of his classic 200-system range. The original Buchla module is still highly sought after and has been the inspiration for a number newer modules over the years, the **Doepfer A-101-2** being one of them.

The **Doepfer A-101-2 Vactrol LPG** features vactrol control of its LP and VCA parts, which gives it that desirable woody, organic response revered by many. In the LPG mode the sound becomes more dull as the loudness decreases, a characteristic of all acoustic instruments; the harder you strike, pluck, or bow, the richer the overtone structure. Consequently, the Doepfer A-101-2 Vactrol LPG a very good module for imitating the tonal response of tuned percussion. By exciting the vactrols with short bursts of noise or impulse spikes through the CV inputs, the low pass gate response can be used to produce sounds resembling hand drums, steel drums or marimba.

Several different A-101-2 units were used during measuring process, due to the individuality of each unit. This is the origin of the Vact switch which we added for users of Modular to be able to alter between three different sets of vactrols for different flavors and responses.

The classic "LPG", low pass gate, mode is called L+V/EXT on the hardware and user interface.

User Interface

F/A Cutoff for filter, amp offset for VCA or both at the same time (L+V/EXT mode). Function of this parameter is dependent on position of **Function** switch and **G1** and **G2** inputs.

CV2 Attenuator for incoming **CV2** input. At fully clockwise position, incoming CV at **CV2** input will have full range, in effect the same as **CV1** input.

Lev Attenuator for incoming audio. When set above 5, normal oscilla-

tor or modulator level signals (such as the A-110 VCO) will distort.

Res This knob controls the resonance of the low pass filter. When set above 8, the filter will go into self oscillation when the **Function** switch is set to LP mode (see below).

Function This switch determines the operation mode of the Doepfer A-101-2. When set to LP mode (switch set to the left) only the low pass filter part will be active. In L+V/EXT mode both the filter and the VCA will be active, meaning that audio is fed first through filter then VCA. EXT means that the **G1** and the **G2** inputs both are active (see below). In VCA mode (switch set to the right), only the VCA part of the Doepfer A-101-2 will be active and thus no low pass filtering will take place.

Vact The switch selects between three different types of vactrol responses. F is for fast response times, M is medium and S is for slow response.

Indicators

LED The red LED at the top right reflects the accumulated CV level into the LPG. The LED itself is set in series with the vactrols in the Doepfer A-101-2 Vactrol LPG hardware circuit, thus accurately showing the CV response to the **F/A** knob and the CV inputs.

Inputs

CV in 1 Full range CV input for controlling the cutoff for filter, amp offset for VCA or both at the same time (L+V/EXT mode).

CV in 2 CV input for controlling the cutoff for filter, amp offset for VCA or both at the same time (L+V/EXT mode). The effect of the CV input here is dependent on the **CV2** attenuator knob.

Audio In Main audio input. Connected with the input level knob (**Lev**) that controls the incoming audio level (see above).

G1 External control input for the Doepfer A-101-2 Vactrol LPG operating mode. When the **Function** switch is set to L+V/EXT position, a logic high or low will select operating mode as according to the chart on the panel (see description below).

G2 The other control input for external control of the Doepfer A-101-2 Vactrol LPG operating mode. When the **Function** switch is set to L+V/EXT position, a logic high or low will select operating mode according to the chart on the panel: **G1** set high (above approx. 0.55 V) and **G2** set low (below approx. 0.55 V) will set the Doepfer A-101-2 Vactrol LPG in LP mode. **G1** set

low and **G2** set high will result in VCA mode, and finally both **G1** and **G2** set low will set to the LPG mode L+V/EXT (default).

Note: Both inputs high (G1 and G2) is a forbidden state, which will bypass the signal through the module all together. Also worth noting is that when the Function switch is set to LP or VCA position, the G1 and G2 inputs set high will repel the effect (G2 active in LP mode and G1 active in VCA mode). These are all undocumented features of the original hardware and will produce unpredictable results.

Outputs

Audio Out This is the output of the Doepfer A-101-2 Vactrol LPG, and its output is determined by how the **Function** switch is set. In LP mode only low pass filter is output here, in L+V / EXT mode the signal put through both low pass filter and then the VCA. And when the **Function** switch is set to VCA, only that the signal passed through the VCA will be heard.

In Use

The Doepfer A-101-2 Vactrol LPG has a very organic response due to its vactrol control, which can be used to closely mimic the characteristics of an acoustic instrument. The vactrol non-linearities makes the low pass gate a very good module for imitating the tonal response of tuned percussion. By feeding in short bursts of noise or impulse spikes through the CV inputs, the low pass gate can used to produce sounds resembling hand drums, steel drums or marimba. It can of course also be used as a pretty straight forward low pass filter or just VCA.

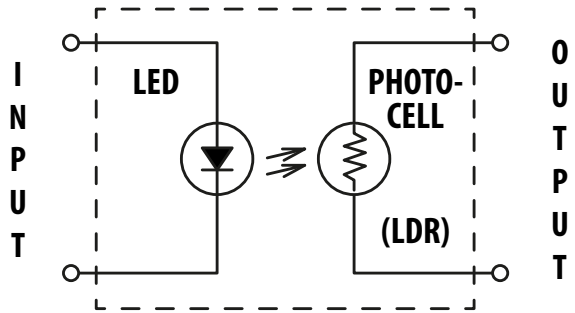
The Doepfer A-101-2 Vactrol LPG also passes and processes DC voltage, which makes it possible to pass a DC offset through and ping the LPG through the CV inputs to get the vactrol response CV on the output.

The Doepfer A-101-2 Vactrol LPG has an input that is quite sensitive in LP mode, so you can use this to distort your signals easily by cranking up the input level.

Use the **G1** and **G2** inputs with the switch in its middle position (L+V/EXT) to sequence the output mode (low pass, LPG or VCA mode) using for example sequencer gates or high outputs from the logic and signal modules.

What the Heck are Vactrols?

Vactrols are a common kind of optical isolator often used in circuitry that requires different circuits to be electronically isolated from each other but still able to



interact. A vactrol consists of a combination of a light dependent resistor (LDR) and a light source (often a LED, light emitting diode) both integrated in a small light-proof case.

The vactrol works by a very simple principle: When input current is applied to the LED element the output LDR turns on and, as the only connection in between the two is light, the output produced is largely dependent on the LDR response. An increase in the LED brightness will cause a decrease in the LDR resistance.

This effect can be used in circuits that require variable resistors to obtain the desired function (for example VCFs, VCAs, VC phasers, VC trigger delays, VC slew limiters, VC Envelope Generators, VC LFOs, VCOs and many more).

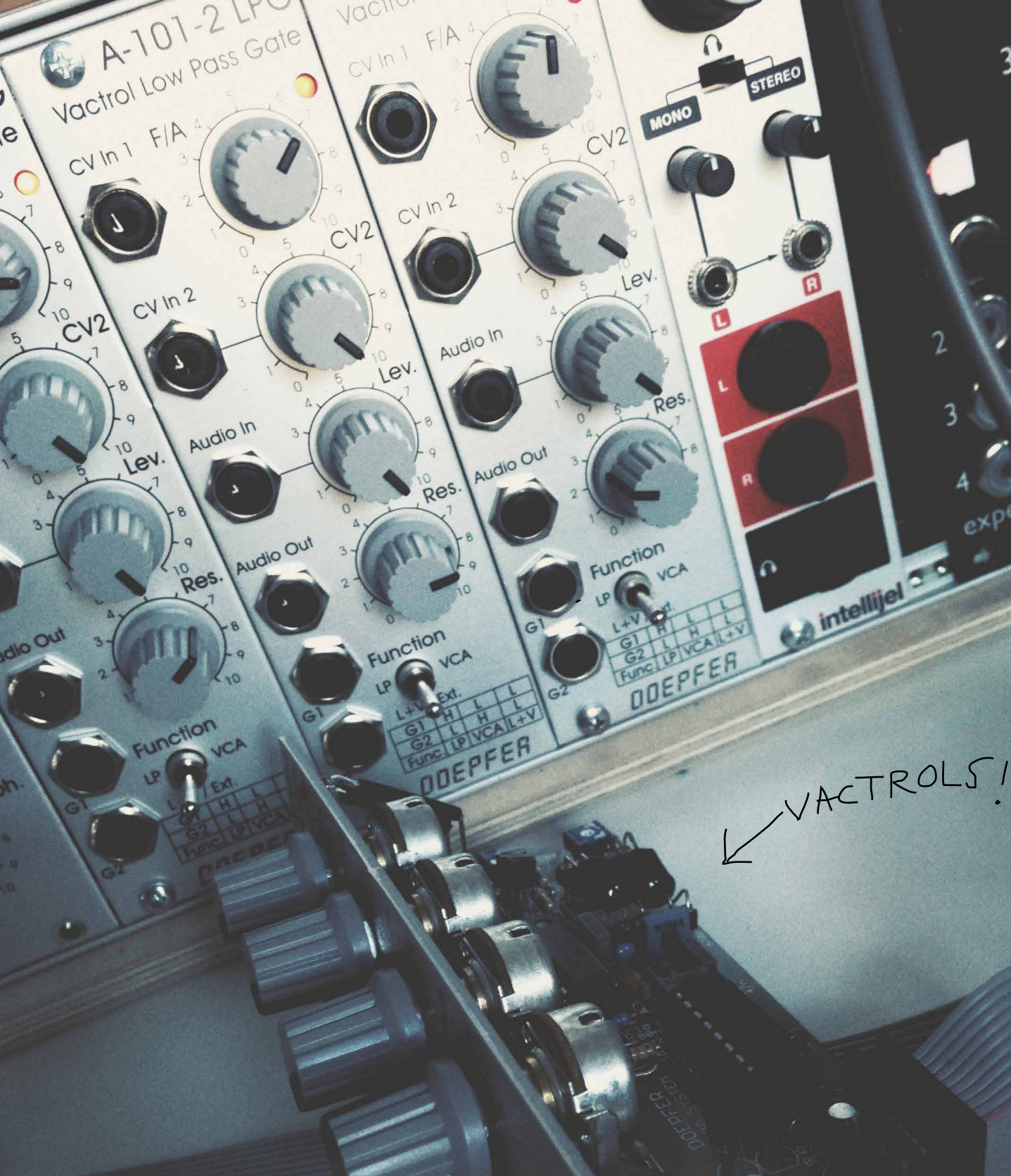
However, mechanical and electrical tolerances will cause each vactrol to behave a bit different from another (for example due to differences in alignment and distance between LED and LDR). This is why identical circuits featuring vactrols will behave or sound a little bit different from each other. Also, another aspect of the vactrol response is the LDR element; that will not respond immediately to the illumination change induced by the LED but will have a certain "sluggishness" built in. The LDR response curve is largely logarithmic and thus it may take a few seconds until the LDR reaches its "dark resistance" (the maximum resistance without illumination). In the upper brightness range it will also respond much faster: up to 50 Hz and more, which corresponds to a response time in the 10 ms range. The essentials of vactrols' behavior will depend very much upon the LDR type and the light-sensitive material used.

As a consequence of the things described above, vactrol based compressor circuits (often called opto compressors) have historically often been considered particularly natural sounding; while a modern VCA compressor can respond almost theoretically perfect with a constant ratio and predictable attack/release curves, many older optical designs have strange attack and release characteristics. In optical compressors, this is originally due to the relatively slow response of a vactrol compared with a transistor-VCA based circuit. This is perhaps the reason why the traditional opto compressors have so much perceived character and "warmth".

The Doepfer LPG response in Modular, coupled with a VCA or another LPG in VCA mode, can yield some really nice sounding results which can be heard among the presets for Modular FX.

Credits

Oscar Öberg – modeling, project management. **Kristofer Ulfves** – presets, validation, user manual. **Henrik Andersson Vogel** – marketing. **Niklas Odelholm** – graphics, validation.


A-101-2 LPG
Vactrol Low Pass Gate
CV In 1
F/A
CV In 2
CV2
Audio In
Lev.
Audio Out
Res.
Function
LP
VCA
Ext.
G1
G2
Func
L+V
MONO
STEREO
L
R
intellijel
DOEPFER
VACTROLS!

18 Weiss MM-1 Mastering Maximizer

Weiss MM-1 Mastering Maximizer features an advanced sound design that can deliver professional-sounding masters with minimum amount of tweaking. "Plugins like the MM-1 are the reason people think mastering is black magic. With very few controls, this plugin somehow just 'knows' how to get your music sounding great – loud, punchy, wide, and the de-esser is incredible! The MM-1 is one of the best mastering tools I've ever used", says Howie Weinberg.

User interface

1. Side menu
2. Input meter
3. Gain reduction meter
4. Output meter
5. Bypass
6. Amount knob
7. Parallel mix
8. Style
9. Limiter gain

Based on Weiss DS1-MK3

Weiss MM-1 Mastering Maximizer is based on technology that is used in Weiss DS1-MK3 – a gold-standard in mastering and TEC award winning mastering compressor/de-esser. Weiss DS1-MK3 is famous for its transparency, flexibility and ergonomics. Weiss MM-1 has all the benefits of DS1-MK3 and takes it to the next level of simplicity, ergonomics and can provide professional mastering results in no time.

Meters

Input Meter Shows Peak and RMS level of the incoming signal.

Gain Reduction Meter Shows actual gain reduction for Left and Right channels or M and S channels (in Wide and Deess styles).

Output Meter Shows the Peak and RMS level of of the output signal.

Parameters

Amount Knob Sets the amount of processing defined by the Style. If you set the Amount to 0% there will be no processing applied before the brick-wall limiter.

Limiter Gain The gain after the initial processing that drives the brick-wall limiter.

Parallel Mix A DRY/WET mix knob for the amount of processing (doesn't affect brick-wall limiter). If turn parallel mix to the Dry side, it will add unprocessed signal to the processed one. At 50/50 position it will be a 50/50 blend between processed and unprocessed signal. For more information about Parallel processing, please navigate to Weiss DS1-MK3 page in the manual and check Bob Katz' article about the technique.

Side Menu Located at the top left corner. You can access Output Trim setting in it.

Output Trim Allows to trim the output for different media. Use industry standards -0.1, -0.3 or -1dbfs (for iTunes Mastering) or custom settings using the continuous Output Trim slider.

Styles

Style is a mastering processing style that is added before the final brick-wall limiter. There are five common mastering scenarios or Styles available in MM-1:

Transparent	Style is designed to be less invasive among the 5 styles.
Loud	Style is designed to deliver high RMS masters.
Punch	Style is designed to add "punch" and "weight" to the masters.
Wide	Style will make the master wider and more "3D". Please note that the style uses MS processing and it's not compatible with mono or multi-mono versions of MM-1.
Deess	Style is a mastering all-round deessing MS algorithm. Will shine during mastering and also mixing applications.

In use

The best way to start mastering with MM-1 is to choose the Style you'd like to use and move the Amount knob to add the amount of processing needed. Watch the Gain Reduction meter in order to see where the processing kicks in. You may want to switch between the Styles and see which one works best for your program material. As soon as you are pleased with the processing Style and the amount, increase the Limiter Gain in order to achieve the desired loudness level.

Tip: You can use several instances of MM-1 in series on one track, as you might want to combine Punch and Wide for example. Please note that only the last instance in the chain should increase the volume with Limiter Gain in order to avoid possible distortion.

Credits

Paul Shyrinskykh – sound design and product management. **Arvid Rosén** – DSP and framework programming. **Björn Rödseth, Kim Larsson** – framework programming. **Niklas Odelholm** – graphic design. **Ulf Ekelöf** – 3D rendering. **Daniel Delviken** - marketing. **Igor Miná** – user manual layout.

19 Drawmer 1973 Multi-Band Compressor

MORE CONTROL, MORE VERSATILITY, more definition—the **Drawmer 1973 Multi-Band Compressor** plug-in lets you micro-manage the dynamics of your sound with ultimate precision. Make your mixes wider and louder, shoehorn a piano into a dense mix, or curb the temper of an unruly bass guitar. And do it all with the unmistakable analog feel of a modern Drawmer classic.

Carefully modelled on the original Drawmer 1973 hardware in collaboration with the Drawmer research and development team, and approved by **Ivor Drawmer** himself, the Drawmer 1973 brings the award-winning three-band compressor bang up to date.

Mastering with the Drawmer 1973 gives you independent control over three frequency bands, often a miracle-cure for problem mixes. Pull out individual instruments, brighten the mix, or tighten the lows … or of course do it all! But the Drawmer 1973 is equally useful on single sound sources in a mix. Take away the boomy character of an acoustic guitar in a perfectly natural sounding manner. Control the low frequency dynamics of a bass but leave its top end alive and kicking. Or remove sharp consonant sounds from a vocal track without eliminating detail and clarity.

Staying faithful to the look and sound of the original Drawmer 1973, we've added some extra up-to-date features to the plug-in, including **Mid/Side** processing and sidechain, giving even more versatile compression and dynamic control. So the Drawmer 1973 plug-in lets you play with the mid portion of a sound without affecting the rest of that frequency, or link the gain reduction to an external sound source.

Low Band Mid Band High Band

User Interface

The Drawmer 1973 panel consists of four different sections, an Input/Output section that handles input and output volumes and signal routing. The lower section consists of three different frequency bands with similar, but not identical functionality. Here follows an overview of all controls, from left to right.

Input/Output Section

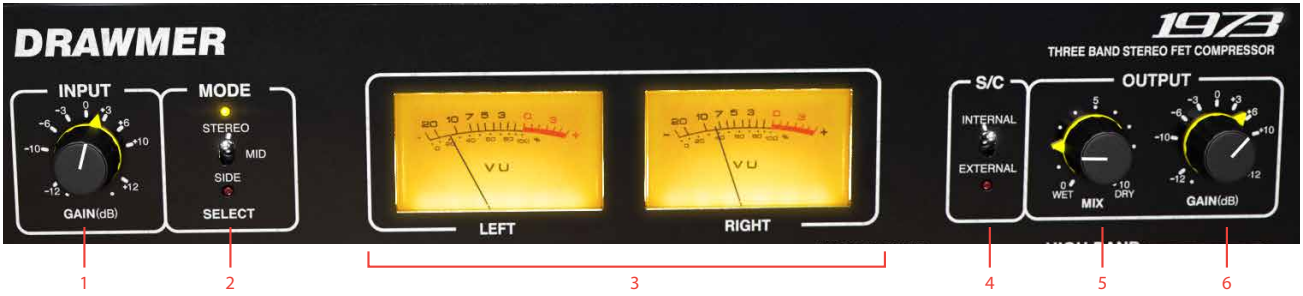
Input Gain Controls the gain going into the unit.

Mode Sets the 1973 in either STEREO or MID/SIDE modes—see a further explanation on using MID/SIDE below. The unit is in MID/SIDE mode in both MID and SIDE positions. Selecting MID displays the controls of the MID portion of the sound, selecting SIDE displays the controls of the SIDE portion of the sound.

Please note that when you switch from STEREO mode to SIDE mode, the (yellow) knob settings from STEREO mode will get copied to the (red) SIDE mode. Therefore, all changes made in SIDE mode will get overwritten if you switch to STEREO from SIDE mode.

If you have made adjustments with the Mid or Side setting selected and then switch back to Stereo, the settings made in the Side mode will be lost.

Input/Output Section



1. Input gain
2. Mode Stereo/Mid/Side
3. VU Meter
4. S/C Internal/External
5. Mix
6. Output gain

VU Meter shows the output level for the left and right channels.

S/C Sets the sidechain mode. When set to INTERNAL, the 1973 operates in normal mode which means that the compression circuits respond to the incoming signal. If set to EXTERNAL, the 1973 will use an external sound source in the detector circuit. For example, external sidechaining can be used to get an entire mix to "duck" with every hit of a kick drum.

If **Sidechain** set to External without external signal routed to the 1973, it will operate as in Internal mode.

Please note that when in Mid/Side mode, sidechain signal will be split to Mid/Side as well. Mid part will trigger the Mid channel of the 1973, and Side part will trigger compression of the Side channel.

Mix Controls the relative level between dry signal (unprocessed) and wet signal (processed by the 1973's circuits). **Mix** is useful for easily setting up parallel compression, where a certain amount of the dry, unprocessed sound is mixed into the compressed sound. This achieves a sound where you get the dynamic control and coherence from the processed sound, but still keep a dynamic feel from the original sound source.

Output Gain Controls the output gain from the unit.

1. Big
2. Low X-Over freq
3. Threshold
4. Status
5. High X-Over Freq
6. Air
7. Attack
8. Release
9. Output gain

Frequency Bands

Between the frequency bands sits the knobs that control the crossover frequency of the bands.

Low X-Over Freq and High X-Over Freq Selects the crossover frequencies for the 6 dB per octave crossover filters. These are the frequency points where one band stops to process audio, and another band takes over. **Low X-Over Frequency** (50 Hz - 1.3 kHz) sets the frequency point at which the split between LOW and MID bands occurs. **High X-Over Frequency** (1 kHz - 14 kHz) sets the frequency point at which the split between MID and HIGH bands occurs.

Input and Output Gain, Mix and X-Over Freq knobs are global and affect both Stereo and Mid/Side modes.

Threshold Determines the signal level above which compression starts to take place. A low **Threshold** setting means more of the incoming signal gets processed. The 1973 is a soft knee compressor, which means that more compression is applied gradually as the input signal level increases.

Attack Controls how quickly the compressor responds to signals that exceed the threshold level. All times are nominal, the actual attack time is further modified by the release setting.

Release Adjusts how quickly the compressor ceases to process the sound after the input level has fallen below the threshold level. The first three switch positions are fixed and provide progressively increasing release times, while PGM DEP ("program dependent") positions F, M and S ("fast", "mid" and "slow") cause the release times to vary in a manner which automatically adapts to the dynamics of the incoming signal.

Gain Turning this up compensates for the gain reduction which occurs when the compressor is active. As the Drawmer 1973 is a multi-band compressor, the three gain controls can be used to achieve subtle or radical changes to the overall equalization curve.

Hold Shift on your computer keyboard while adjusting a setting to link the corresponding control in the separate bands.

Gain Reduction meters The meters operate in -1,-2,-3,-5,-7,-10,-15,-20 dB range.

Status Activates (ON), bypasses (BYPASS) or mutes (MUTE) the individual bands.

All the parameters above are common for all frequency bands. The controls below are specific for the low and high band respectively.

Big The Big circuit makes the low frequencies appear louder or "bigger", by reducing the sidechain's sensitivity to low frequencies. The BIG mode is useful on a stereo buss or in a mastering situation, where you want to control the dynamics but still keep a thick and warm low end tone. It is only available for the LOW band.

Air Air can be used to reintroduce high frequencies, which can sometimes be lost as a result of heavy compression. Activating **Air** makes the sound more intimate, detailed and transparent, but without making it sound harsh or introducing any noticeably unnatural artifacts. **Air** is only available for the **High** band.

In Use

The Drawmer 1973 Multi-Band Compressor splits the frequency spectrum into three bands—Low, Mid and High—and lets you add compression to these individually. This is useful in a number of cases. Here are some examples.

Controlling a Boomy Guitar or Piano

Sometimes, you will encounter a recording of an instrument where some notes seem to stand out in an annoying manner. Perhaps there is an acoustic resonance in the recording room, or in the body of a piano or an acoustic guitar, which is excited when certain notes are played. So every time a particular range of notes are played, the resonance just seems to explode. A similar situation often occurs with rhythm guitars in heavy metal, when the player goes back and forth between letting power chords ring out and "chug-chug" style palm muted playing. The palm muted sections will often have an overbearing bass boost, which you might want to control in the mix. In cases like these, simply cutting the offending frequency with an equalizer isn't the best option, as this affects that frequency at all times.

But with the Drawmer 1973, you can use one of the bands to zoom in on the disturbing resonant frequency and compress it whenever it gets too loud—and only then. First, decide if the LOW, MID or HIGH band is the suitable choice for your application. Set the other two bands in MUTE mode (with the **Status** switch), and adjust the crossover filter(s) until you have found the specific frequency range you want to control. Set the **Threshold** low enough for it to compress every time the frequency stands out too much, but high enough to leave the frequency unaffected at all other times. When you obtain a nice and even response, engage the two other bands again. Either set the 1973 to BYPASS them, or adjust their parameters as you see fit.

Create a Solid Bass Track

For very dynamic bass guitar playing styles, such as a funky slap style or a hard rocking picked bass, you may find yourself wanting to keep the low end of the bass tight, even and controlled without strangling the life and attack that you'll mostly find in the midrange and treble with playing styles like these. Here, you can use the Drawmer 1973's LOW band to do precisely that—smooth out the dynamics of the low frequencies to create a solid ground for the mix, but use lighter compressor settings for the MID and HIGH bands, or bypass them entirely.

Compress a Subgroup or the Master Buss in a Transparent Manner

With a full-band compressor, you will find that every time a part of the sound source exceeds the threshold level, the entire output gain will be reduced. This may be just the effect you are after, but in other cases, Drawmer 1973 can be used to create a transparent dynamics control. This is because none of the three bands will react to what happens in any of the others. So for example, a loud bass drum hit will not make the cymbals duck, as they would with a full-band compressor. This means you can control the dynamics, but in a less apparent manner.

Adjust Specific Sound Sources in a Stereo Mix

The Drawmer 1973 plug-in offers a mode which is not available on the hardware—the MID/SIDE mode (MS). In MID/SIDE mode, the Drawmer 1973 will not process the left and right channels separately. Instead, the sound is separated into the MID and SIDE portions of the sound. MID contains only the material which is common to both left and right channels (ie. the mono portion of the sound). This should not be confused with the MID frequency band. SIDE contains only the material which differs from left and right.

MID/SIDE processing is typically useful in a mastering situation, where you are working on stereo program material without access to the individual tracks. Perhaps you will find that a particular sound source—such as the lead vocal—has been mixed too low, and you want to bring it forward. But since you don't have access to the individual tracks, you can't just raise the lead vocal's level. You could apply EQ to bring up the frequencies that are most dominant to the vocal, but you would then also bring up all other instruments in that frequency range.

In this case, you could use the Drawmer 1973 in MID/SIDE mode. The vocal will typically be mixed dead center, so it will only be present in the MID channel. Therefore, you can select the MID channel with the **Mode** switch, and use its crossover filters to have one of the 1973's bands zoom in on the frequencies where the vocal is prominent. Boost the output of this band, and you will find that you have brought the vocal forward without affecting the chord instruments such as guitars and keyboards by much, even though these exist in the same frequency range as the vocal. Because these will typically be either panned left or right in the stereo panorama, or stereo recorded. That means they mostly reside in the SIDE channel and are thus not affected by the changes you made to the MID channel, at least not by as much as the lead vocal was.

Widen the Mix

MID/SIDE processing can also be used to make a stereo sound appear wider. You do this by slightly turning down the MID channel relative to the SIDE channel. But be careful not to go too far with this, as you might end up with a center "hole" in the mix. You may find that turning down the HIGH band of the MID channel, but leaving the LOW and MID bands as they are, gives a widening effect but still retains the integrity of the mix.

External Sidechaining

The typical example for sidechaining is to get an entire mix to "duck" every time a kick drum hits—this is a very common technique in electronic dance music. In order to achieve this, you set the 1973's **S/C** switch to EXTERNAL, and select which sound source should be routed into the sidechain of the 1973 (please refer to your DAW's manual for the specifics on how to do this). With this in place, the 1973 will start reacting to the sound of the external sound source that is fed into the sidechain. So in the above example, this sound source would be a kick drum. Every time the kick drum level exceeds the threshold level, the 1973 will react and start compressing.

Original Drawmer 1973 hardware.

External Sidechain in Mid/Side Mode

If you set the 1973 to MID/SIDE mode and use external sidechaining, the sidechain is actually also set to MID/SIDE operation. This means that you can use the external sound source to control only the MID or SIDE portions of the sound. For example, you might find that the kick drum and bass instrument struggle for the same space in a mix. In this case, you could feed the kick drum into the sidechain, use the Mid channel to zoom in on the low frequencies of the bass instrument (which is typically mixed dead center), and set the **Threshold** so the bass instrument ducks every time the kick drum hits. You will now have the bass instrument "wrapping around" the kick drum instead of conflicting. But everything else in the mix will be unaffected.

Parallel Compression

In any of the above use cases, you can use the **Mix** knob to apply parallel compression. This is a very useful technique in which you create a blend of the processed and unprocessed sound. The processed portion will add a full and dense sound, while the unprocessed portion retains the original dynamics.

Keyboard Shortcuts

SHIFT-click a parameter to adjust it for all bands at the same time. If you SHIFT-click the "status" label, that band will be soloed, ie. the other bands will be muted.

Shift-click to Solo the current band.

The fastest way to un-solo (un-mute the other two bands) is to shift-click **On**, which will turn on all three bands.

Setup Options

The normal operation for the Drawmer 1973 is to copy

the Stereo parameters over to the Side parameters, whenever you go from Stereo to Mid/Side. It is possible to turn off this option by disable "COPY STEREO PARAMETERS TO SIDE" in the Setup menu.

Credits

Björn Rödseth, Oscar Öberg – modeling. **Paul Shyrinskykh** – product owner, manual. **Niklas Odelholm** – graphic design. **Kristofer Ulfves** – marketing. **Patrik Holmström** – framework programming, **Henrik Andersson Vogel** – additional manual copy. **Ulf Ekelöf** – 3D rendering.

20 Drawmer S73

FOR THE MASTERING ENGINEER, a multi-band compressor is a powerful tool. But as Spiderman's uncle Ben famously says: with great power comes great responsibility. A multi-band compressor that is not used diligently may create more problems than it solves. This is why we created Drawmer S73 Intelligent Master Processor.

Drawmer S73 Intelligent Master Processor features an enhanced multi-band compressor sound design, fine-tuned to improve your mix with a minimum of tweaking. The S73 does most of the job for you by making the choices a mastering engineer would make. Simply use the **Style** parameter to switch between ready made mastering processing techniques to quickly find the sound that suits your mix the best.

Original Drawmer 1973 hardware.

Handles It All

When we designed Drawmer S73 IMP, we wanted to make it sound transparent and suit any program material. We analyzed dozens of mixes and reached the conclusion that most of them have similar problems. So we created algorithms for S73 IMP to fix those problems. S73 can deliver both gentle sheen for a mix that's already great to begin with, as well as dramatic improvements for a more problematic mix.

Based on Drawmer 1973

Under the hood, the S73 is based on technology created by Drawmer for the **1973 Three Band FET Stereo Compressor**. This is an award winning multi-band analog compressor design famous for its precision and flexibility. Softube has carefully modeled all the characteristics of the original analog hardware, and have taken the concept a step further by adding mastering sound design to create the Drawmer S73 IMP. The unit offers the perfect way to balance and shape your mix, and get great analog sound in no time.

1. VU meters
2. Air
3. Style
4. Amount
5. Gain reduction meter
6. Mix
7. Gain

User Interface

Amount Controls the amount of dynamic and spatial processing applied. However, you should use **Mix** knob to control the overall precessing level.

Gain reduction meter Shows the overall gain reduction

Style

Switches between the different styles of mastering algorithms. The algorithms are divided into three groups: CLEAN, BASS CONTROL and SPATIAL.

CLEAN group contains styles that make a mix clearer by improving definition between vocals, instruments and other elements of the mix. Scroll through the styles in the group to find one that suits your music the best.

BASS CONTROL group lists styles that treat problems in the low end. If your mix sounds boomy or muddy, try De-Rumble style first. Punch style will add punch and definition to the kick drum and bass.

SPATIAL group includes styles designed to widen stereo image of your mixes, bring out ambience and emphasize lead vocals in the mix.

VU meters	VU meters show output volume for left and right channel.
Mix	Mix knob controls the balance between DRY and WET signal and can be used for parallel processing.
Gain	Gain knob controls the output gain.
Air	**Air** switch will bring more "air" to your mixes. Use MIX knob to control the amount of **Air** you're adding to the mix.

In use

Using the Drawmer S73 is easy, since you really can't do anything wrong. Play around with different styles, place several instances after each other, and listen to the result.

Select style

Fundamentally, the Drawmer S73 IMP is operated by auditioning the different **Styles** and choose the one that suits your mix the best. It's good to start with 1-3 dB of gain reduction and increase the amount if needed. It's safe to combine two or more instances of S73 with different styles. Just add them in series in the insert slots of your DAW's master channel. For example, an S73 instance with GENTLE compression often works well followed by one set to WIDE mix. As does DE-RUMBLE followed by CLARITY 1. See the **Styles** description section for a walkthrough of all included **Styles**.

Use parallel processing

The Mix Knob can be used for parallel processing. This is a very useful technique in which you create a blend of the processed and unprocessed sound. The processed portion will add a full and dense sound, while the unprocessed portion retains original dynamics. To use parallel processing, try increasing Amount until meter shows 5-10 dB of gain reduction—which is in most cases is considered a great deal of compression—and then use the Mix knob to blend in the unprocessed signal until you find a pleasing balance. Activating AIR will add even more excitement to the master.

Tip! Use several S73 instances with different **Styles** at the same time, for example De-Rumble followed by Clarity 1.

Styles description

Here is a short description of the different mastering styles.

Clean

Clarity 1	Instant clarity and definition. Use gently.
Clarity 2	Adds definition in the midrange and tames high frequency harshness.
Neutral	The most neutral Style, ideal to polish mixes that are already great, and just add some sheen.
Gentle Comp	Adds glue and density to the mix. Keep gain reduction around 3 dB.

More Air	Lets your mix breathe, and adds sheen to the top end. This style allows you to control the amount of air by using the **Amount** knob, while the **Air** switch will add additional sheen.

Bass Control

De-Rumble	Removes low frequency rumble from the mix and adds clarity and definition.
Punch	Adds punch to a kick drum and bass.

Spatial

Wide Mix	Gently widens your mix in the stereo field.
Ambience	Brings out existing ambience in your mix.
Vocals Enhance	Will bring the lead vocal upfront.

Please note that Spatial style group is not compatible with mono and multi-mono versions of S73

Credits

Björn Rödseth, Oscar Öberg – modeling. **Paul Shyrinskykh** – sound design, manual. **Torsten Gatu** – project management. **Niklas Odelholm** – graphic design. **Kristofer Ulfves** – marketing. **Ulf Ekelöf** – 3D rendering.

21 Eden WT-800 Bass Amplifier

MODERN MUSIC PRODUCTION IS ALL ABOUT THE BASS. Whether it's a Moog synth line, a throbbing EDM rhythm or a poetic fretless bass, you need a solid and clean low end. The Eden WT-800 Bass Amplifier is the ultimate bass recording tool for producers and bass players alike.

Eden's flagship bass amplifier **WT-800** is heralded as one of the all-time great bass amps. Every detail has been fine-tuned for bass. The WT-800 plug-in does not only deliver the powerful and present sound of Eden's high-end bass amp. It also includes models of Eden's top of the line 4x10" and 1x15" cabinets, mic'd with four high-end studio microphones each. The mics were selected and positioned by producer and bass player **Troy Antunes**, known for his work with **Justin Timberlake, NSYNC** and **Al**

Green. Troy Antunes has also provided some expertly designed presets. Unlike vintage style bass amps, Eden WT-800 delivers a modern, dynamic, and rock solid bass sound, flexible enough to be used for R'n'B, hip-hop, metal or jazz.

Eden WT-800 extends beyond a bass amp model—it's a fully featured tool for any bass instrument recording. Don't limit yourself to only using it with bass guitars—it adds weight and focus to synth basses as well.

High fidelity for low frequency.

Simple as that.

User Interface

If you are familiar with Softube's Marshall plug-ins, you'll recognize the hidden channel strip panel at the right. Click and drag the amp to the left, or click on the panel edge to reveal the channel strip.

Amplifier Parameters

Gain Adjust input gain to the amplifier. Set it so that the green light is blinking when you're playing.

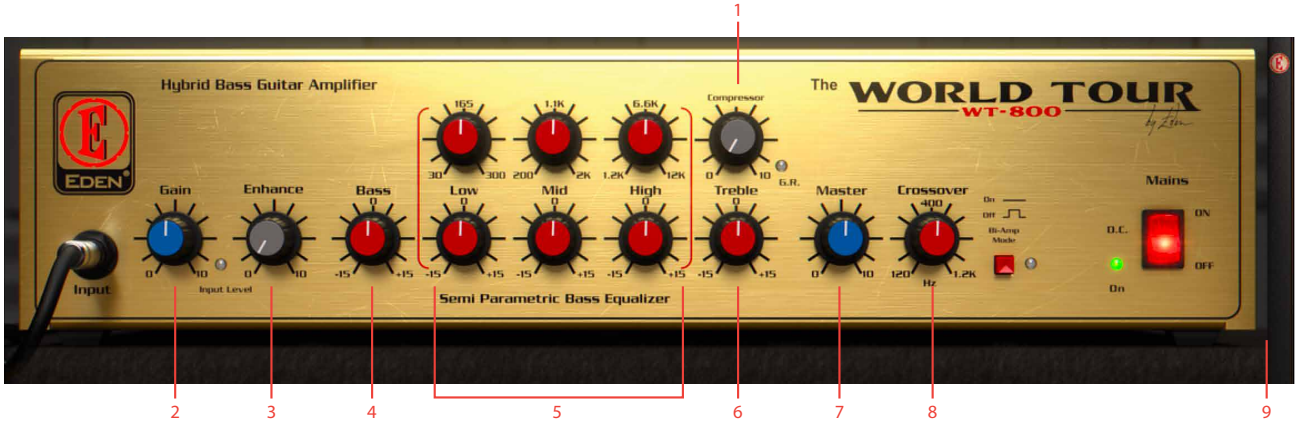
Enhance The magic of Eden. This is a modern bass sound with the twist of a single knob. It's usually the first thing you adjust.

Bass, Treble If you find a three band parametric EQ a bit much for a bass amp, start with the **Bass** and **Treble**. They work just as you would expect.

The **Bass** is a type of shelving filter at around 100 Hz. **Treble** is a high shelving with center frequency at 4 kHz.

Low Gain/Frequency
Mid Gain/Frequency
High Gain/Frequency A three band parametric equalizer; enhance the mid, tailor the highs or tame the low end.

Compressor One knob compressor with a punchy attack and fat release. This compressor will **add more punch and attack to your tone**, rather than destroy the transients. If you play with a soft touch you'll get a looong sustain with a bit of compressor dialed in. The **Input Gain** will also affect the amount of compression, so a higher input gain will lead to more compression.



1. Compressor
2. Gain
3. Gain
4. Bass
5. Low gain/Frequency
   Mid gain/Frequency
   High gain/Frequency
6. Treble
7. Master
8. Crossover, Bi-Amp mode
9. Open channel strip

G.R. LED The gain reduction LED lights up when the compressor is limiting the output volume.

Master Main volume. Use this as the primary, overall, output volume knob.

Crossover, Bi-Amp Mode In **Bi-Amp Mode**, the output from the amp is split up into a low frequency and a high frequency part with the crossover frequency set by the **Crossover** knob. The low frequency signal is fed to the 1x15" cabinet, and the high frequency signal to the 4x10". With **Bi-Amp Mode** engaged, you can get a very clear and solid mid-range.

1. Mic selection 1x15
2. Mic selection 4x10
3. Tweeter level 4x10
4. Clip LED
5. Phase invert
6. On/Off
7. Level and channel meters

Channel Strip Parameters

The channel strip consists of three channels: the mic on the bassy 1x15" cabinet, the mic in front of the brighter and more well-balanced 4x10" cabinet as well as audio direct from the amplifier. This is not the signal from one of the original WT-800's built-in direct outputs, but an output taken from the end of the amplifier; after the **Master Volume**.

Mic Selection (only 1x15" and 4x10" channels) Both the 1x15" and 4x10" cabinet has a selection of four different mics. Select which mic to use with these two columns with four radio buttons. For more information regarding the microphones, see the section below.

Phase Invert (all channels) Phase inverts the audio on that channel. Keep in mind, there is no "correct" phase when it comes to bass. Flip the phase and see what happens. If you like the sound, keep it, otherwise flip it again.

In (all channels) Turn on the audio on that channel.

Level (all channels)	Set the level of the respective channel.
Tweeter Level (only 4x10" channel)	Turn up for more high frequencies from the 4x10" cabinet. The 4x10" has a built-in tweeter, with a level control on the back. This knob adjusts that level.
Meters, Clip LED	The channel meters show the peak levels of each channel. The **Clip LED** at the top right lights up when the output signal exceeds 0 dB.

Cabinets and Microphones

How do I get a clear and uncolored tone with lots of headroom? That was the question that started Eden's quest for the perfect bass cab. They started building bass cabinets in the late 70s and their designs quickly became an industry standard for touring musicians. Since you didn't have to trade low end with clarity anymore, it's no wonder that it became a success.

The cabinets we chose to include in the Eden WT-800 plug-in are the cabinets most associated with that amp: the **D410XLT** and **D115XLT**.

The 4x10"

The **D410XLT** has a 4x10" speaker layout and dual ported design, and offers focused tone with exceptional low end. It is the perfect option when you need to project your sound through the mix and drive for ultimate lows.

Troy Antunes during the Kore Studios sessions.

The 1x15"

The **D115XLT** is a tuned and ported professional cabinet with a fast responding 15" speaker. It offers a classic rounded tone with great detail and response, coupled with a high temperature voice coil and our Eden cast bell tweeter. It's the option for players looking for a little more growl.

Microphones

Put three bass players in a room and there will be heated and long discussion about mics. But thanks to moderating non-bass players we managed to put the differences aside and come up with a roster of five diverse and very different sounding mics.

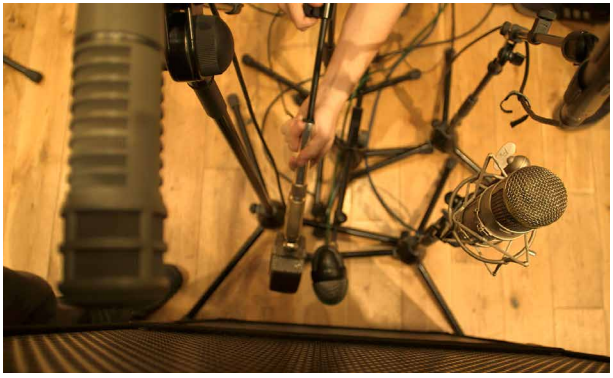
The **AKG D112** has a classic scooped-out sound that just sounds right. Might feel a bit boring, but it does the job. **Electro Voice's RE20** is another classic, with not as much of the lowest and highest as the D-112, but a focused and warm midrange. **Neumann U 47** just has to be included. Full range, sparkling highs, nice and balanced. Can't go wrong there.

The final two mics are not studio classics for bass, but completes the palette in a very nice way. Royer's ribbon **R-121** has a super smooth top, and tames the 4x10" like nothing else. It's sounds both aggressive and smooth at the same time (a passive/aggressive mic?). And last, but definitely not least: the huge **Melodium Type R** ribbon mic is as big as it sounds. Nothing sounds like that one.

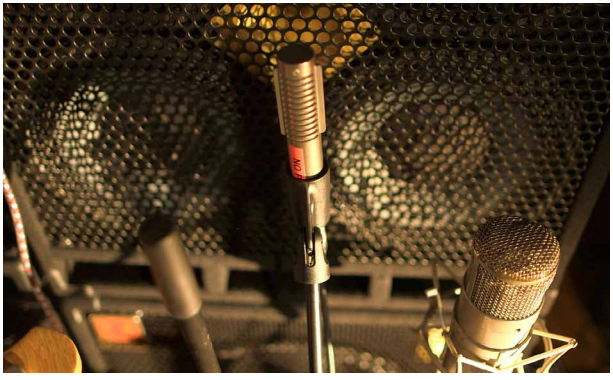
George Apsion, owner of Kore Studios, and lucky owner of that particular mic says: *"The Type R is a secret weapon for bass. It's an old French ribbon mic dating from the early 1940s. We were lucky enough to find one with the original ribbon intact and it has a huge low bump in the frequency response. There can't be many left out there, so it's a great feeling to hear it producing such immense sounds in a modern setting."*

All cabinets and mics were recorded at Kore Studios, London, UK, by Troy Antunes, Niklas Odelholm, Kim Larsson and George Apsion.

Melodium Type R



Setting up mics for the session.



The Royer R-121 and U 47

The real deal.

Meters

The peak meters in the channel strip shows the peak level of each individual loudspeaker or direct from the amp. Even if all three meters are below 0 dB, that won't guarantee that the combined level is below 0 dB, since the sum of all can be higher. If the **Clip LED** at the top right corner of the channel strip is lit, you'll definitely above 0 dB. Please note that the Eden WT-800 plug-in won't limit the audio level at 0 dB, but can output much louder audio than that. So if you have a DAW that allows levels above 0 dB, you don't have to bother too much about the **Clip LED**.

Mono and Stereo Operation

The Eden WT-800 is in its most basic form a mono plug-in, but if you place a stereo instance of the Eden WT-800 on a track, it will process both channels independently. You can thus use the stereo version of Eden WT-800 as an effects processor on stereo sources, such as a stereo synth bass line or drum kit.

Presets

Many of the presets have a carefully tuned equalizer, sometimes to match a bass player, style, song or tonestack of a vintage amp. When adjusting presets, reach first for the **Bass** and **Treble**, **Compressor**, **Enhance**, cabinets, and lastly the semi-parametric equalizer. **Compressor** settings depends a lot on your input gain and bass, so it will always be necessary to adjust the amount of compression to your taste.

Credits

Kim Larsson – modeling, programming. **Niklas Odelholm** – project management, sound design, graphic design, user manual. **Arvid Rosén** – extremely complicated Sharc related stuff. **Björn Rödseth** – programming. **Patrik Holmström** – programming. **Paul Shyrinskykh** – project management, quality assurance. **Troy Antunes** – sound design, presets. **Jerker Odelholm** – presets. **Ulf "Rockis" Ivarsson** – presets.**Ulf Ekelöf** – 3D modeling.

22 FET Compressor

Introduction

THE FET COMPRESSOR IS BASED ON the most famous hardware FET compressor, and very much effort has been spent modeling the real hardware to make sure that this one sounds exactly the same. If you only use the big knobs and the six fixed ratios you'll get exactly the same analog sound and functionality as the real deal. But that's just the beginning. With the help of digital technology we have added some useful features that still have that analogue sound – but features that would've been hard or even impossible to implement with analog electronics. That way you will be able to get the best of two worlds. (Not even mentioning how ridiculously many FET Compressors you will be able to fit into your project without running low on CPU.)

Not convinced yet? Set the **Input** on MAX, **Ratio** on ALL. Listen.

Design Philosophy

THERE ARE THREE MAIN IDEAS behind this design: first of all, we wanted to make an analog sounding digital compressor. Nothing must stand in the way of the sound. Every single part of the simulation contributes to making this piece unique. The extremely fast attack, all the subtle (and sometimes not so subtle) distortion that comes from the different parts of the compressor and the extremely careful way the signal is handled and conditioned through out the signal chain.

SECOND, WE ADDED FEATURES to make this product even more versatile and unique. The enhanced ratio, parallel compression, detector filtering and lookahead are such features. Third, these features must not stand in the way of the usability. The fewer knobs the better, and the faster the user gets the sound they want, the better.

This boils down to a single main objective: Every user should be able to get a good sound within seconds, and no user should be afraid to mess up the sound. An amateur should be able to make this sound just as good as a pro. And yeah, it has to look good.

In order to achieve these objectives we had to put special effort into the modeling. The original hardware has some quite quirky circuits, and a lot of its sound comes from far from ideal components and design. A lot of new ideas were developed during the modeling, and with the help of our expert listener and "golden ears" Dan Lumbye, we managed to put together a piece of software that should please even the most critical ears.

1. Input volume
2. Input VU
3. Gain reduction VU
4. Parallel inject
5. Output volume
6. Output peak meter
7. Ratio
8. Attack time
9. Release time
10. External sidechain volume
11. Low cut
12. High cut
13. Lookahead

User Interface

THE CONTROLS OF THE FET COMPRESSOR are divided into two sections.

First of all we have the big knobs (**Input**, **Ratio**, **Attack**, **Release** and **Output**), which are the knobs that you will use every time you use the compressor. These knobs are pretty standard and you probably already know what they do.

Then we have the smaller knobs (**External Sidechain**, **Low Cut**, **High Cut**, **Lookahead**, and **Parallel Inject**), which don't have to be tweaked every time you use the compressor. You don't even have to feel bad if you never touch them. People have managed to make hit records for 40 years without them. Nevertheless, they are extremely useful, and soon you'll have a hard time understanding how you ever could've managed without them.

In addition to this, we also added some metering so that you can monitor your signal all the time. An input VU meter (which is fun to drive into the reds all the time) and a stereo output peak meter make sure that you always know what is going on with your signal.

Input

The **Input** knob on this compressor works both as a gain control and a threshold control. If you increase the **Input** volume you will also increase the gain reduction, which makes it work sort of like a combination of a gain knob and a reversed threshold knob. You can drive the INPUT VU far into the reds without any unwanted digital clipping. Just make sure that the OUTPUT PEAK meter doesn't indicate any clipping. (If it does, just lower the **Output** volume a bit.)

> The **Input** knob also works as a threshold control. More input gain will give you more gain reduction.

If you drive the input volume so that the VU start hitting the reds you will also add more distortion to the signal. If you want lots of distortion we recommend you to also set the **Ratio** knob on ALL.

> More input gain also means more distortion (in a good way).

Ratio

At a first glance, the **Ratio** knob on the FET Compressor seems to work a little bit different than its hardware counterpart. The biggest difference is that the **Ratio** knob is continuous, meaning that it's possible to select settings between different fixed ratios, for instance between 12:1 and 20:1. Furthermore, it's also possible to select settings between 20:1 and the famous "ALL BUTTONS IN" setting.

If you would rather use the "predefined" settings that are identical to those on the original hardware: 1:1, 4:1, 8:1, 12:1, 20:1 and ALL, just click on the labels to jump directly to those settings.

"All Buttons In" Mode

The ALL mode is a peculiarity found on this type of compressor. Originally the different ratios were selected with push buttons, which made some mix engineers try out what happens if you press all buttons at once. From a technical perspective, pushing all buttons at the same time makes all bias levels go berserk. From a sound perspective, this means that you will get a very pumping compression with lots of distortion. This mode is often called the "British mode". By setting the **Ratio** somewhere between 20:1 and ALL you'll get a sound that's definitely unique for this product.

Attack and Release

Use the **Attack** and **Release** controls to change how fast the compressor starts to compress (**Attack**) and how fast it should recover from the gain reduction (**Release**). The attack time on this unit is very fast, ranging from about 20 µs at the fastest setting to about 800 µs (that's *micro*-seconds!) on its slowest setting. Other kinds of compressors often have much slower attack times. In comparison to the attack time the release times are much slower – ranging between 50 ms and 1.1 s. Please note that these numbers only give a general idea of the attack and release times. In practice (and just like in the modeled hardware), the attack and release times will be program dependent, ie. depend on the characteristics of the input signal.

> Since a fast attack and release time leads to a fast gain reduction, sometimes those settings will cause unwanted "clipping effects". To remove this unwanted side effect, you can either slow down the release time or add some look ahead to the detector circuit. (See "Lookahead").

Input and Gain Reduction VU meters

The **Input VU** meter is used to monitor the input signal. If a stereo signal is present, the meter will display the maximum energy of both signals. This may seem like

a drawback, but since the gain reduction circuit (a.k.a. "detector" or "side chain") works in mono, the VU will actually display the same thing the detector sees. (Unless you start fiddling with the small knobs, but we're not there yet.)

The **Gain Reduction** meter is used to monitor the amount of gain reduction in the compressor. The FET Compressor has only one detector, so if a stereo signal is present the gain reduction will be the same for both channels. This prevents unwanted distortion of the stereo image.

If the **Ratio** knob is set at ALL, the **Gain Reduction** meter won't show the same results as the original hardware. Instead it will display the actual gain reduction (which the original unit doesn't). You will also notice that the **Gain Reduction** sometimes display positive values when the **Ratio** is set at ALL, which is correct. The ALL mode does indeed have negative gain reduction (=positive gain) for some settings.

Output

In comparison to the **Input** knob, the **Output** knob doesn't do much more than just adjusting the output volume. Keep a close look on the OUTPUT PEAK meter while adjusting the output volume to avoid unwanted digital clipping. It is good practice to keep the levels below 0 dB, even if you are using a (native) host that allows level above 0 dB.

Output Peak Meter

The output peak meter is a fast peak meter with hold values. Single peak values up in the reds (at 0 dB) are okay, but don't push it. This meter will display the left and right channel on the left and right meter if you use the plugin in STEREO mode, otherwise it'll just show the same MONO signal on both meters.

Detector Controls

The most important part of any compressor is the detector. It is the detector that decides how the compressor should compress. All controls except the **Output** (and **Parallel Inject**) control the detector, but in this plug-in we have chosen to bundle the four advanced knobs together and call them "detector controls".

Common to all these controls is that they can be turned off by setting them in the OFF position. Their status LED will light up when ever they are active.

> Don't be afraid to keep these controls in the OFF position. Use these controls if there is something particular that you'd like to achieve or something that needs to be fixed. (If it ain't broken, don't fix it!)

External Side Chain

In some plugin formats, such as RTAS, VST3 and AU, it is possible to use an external signal as input to the detector. This is very common if you for instance want to compress a bass line using the bass drum as the trigger to the compressor. In that case, the bass line will be compressed when ever the bass drum is hit.

> Even if an external side chain signal is present, you will have to activate the external side chain by increasing the **External Side Chain** knob until the LED lights up. If the **External Side Chain** knob is in its OFF position, the detector will always use the regular input.

Use the **External Side Chain** knob to set the input volume of the external signal (for instance the bass drum track). A higher input volume will lead to more gain reduction, just like the **Input** knob works without an external side chain.

> Monitor the **Gain Reduction VU Meter** when you set the level of the external side chain. Just increase the **External Side Chain** until you get enough gain reduction.

Low Cut and High Cut

The **Low** and **High Cut** can be used to filter the signal before it reaches the detector. A good practice can be to remove some of the (unwanted) low frequencies so that they don't affect the gain reduction. Another trick is to apply filtering so that only certain frequencies cause the detector to compress the signal. If you for example compress a drum kit, you might want to keep the boominess of the bass drum, but compress the cymbals and snare. Set the **Low Cut** to 200-300 Hz so that the bass drum will get filtered out before it reaches the detector.

Please keep in mind that **Low Cut** and **High Cut** don't alter the direct signal, just the signal that enters the detector.

> Please note that the 200 and 1000 Hz settings on the **Low** and **High Cut** knobs corresponds to the knob being set at 12 o'clock (where the little indicator lines are located). The easiest way to get to that setting is to click on the 200 or 1000 label on the panel.

Lookahead

The **Lookahead** can be used if the fastest attack time isn't fast enough, for instance if you want to apply some heavy limiting to a signal. The **Lookahead** will delay the signal path outside of the detector by up to 1 ms, leaving the detector signal unaffected. This has two effects:

1. The output signal will be delayed by up to 1 millisecond.
2. The detector will "see" the undelayed signal, but reduce the gain on the delayed signal. This means that the detector will be able to compress the signal before the actual transient comes (since the actual transient has been delayed).

> The **Lookahead** time corresponds to the total latency of the plugin. The plug in will not report this latency to the host, since almost no hosts support compensating for a delay that depends on a parameter.

Some very fast **Attack/Release** settings will cause a "crackling" sound on transients, often sounding like some sort of digital clipping or saturation somewhere in the signal chain. This is very typical for this kind of compressor, but is usually an unwanted effect. The normal way to get rid of this "side effect" is to slow down the attack and release times but now you have another tool to use. By adding a little lookahead to the detector, many of these side effects can be avoided.

Parallel Inject

Parallel Compression is a technique where you blend the compressed signal with the original signal. Say that you have a heavily compressed drum track where all the transients have been lost (compressed). By adding a little bit of parallel compression you can blend in the original signal (with the original transients) with the highenergy compressed signal (without transients) and thus get the best of both worlds.

This is a technique that is very easy to achieve with a send bus, but this baby does a little bit more than that. First of all, you won't get any problem with the **Lookahead** since the dry signal is delayed with the same delay time as the compressed signal. Then the dry signal will be affected by the same analogue modeling mojo as the compressed one, which means that some of the distortion and coloring of the compressor will also have a an effect on the dry signal. And at last, the **Output Peak** meter will of course show the sum of these signals, so that you can set a proper output level.

> Whenever you change the gain reduction (for example by changing the **Input** volume) you will need to change the **Parallel Inject** since the volume of the compressed signal has changed. But be careful sometimes the dry signal is much louder than the compressed signal!

Status Indicators

The FET Compressor comes with a couple of handy status labels, located in the VU meter section.

Remote (Pro Tools only)

Whenever automation is used, or if an external control surface is connected, the **Remote** status indicator will light up. The knobs that are being controlled will also get a thin frame around them.

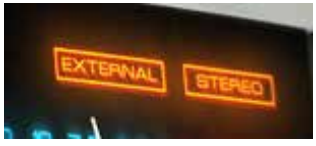
External

This indicator displays when an external side chain is present. Whenever an external side chain is present, it is possible to use the External Side Chain knob to set the amount of gain of the external signal to the detector.

In Pro Tools it will light up whenever you have connected a side chain bus to the plugin. This indicator will always be lit in hosts that use the Audio Unit (AU) format. In VST, it will light up if the host accepts side chain inputs, but this doesn't work for all VST hosts... So it goes.

Stereo

The **Stereo** indicator lights up if the compressor was instantiated in stereo mode.



Tweaking Tips

We didn't want to include too much of the standard "this is how a compressor work" theory, but rather focus on the parts that make this plugin special. Here are our favorite tweaks.

I want distortion!

There are two types of distortion in a compressor like this, first we have the input and output stage distortion and second the (frequency dependent) distortion caused by the compressing. To get the first type of distortion, just increase the **Input** until you get enough distortion. You can even have **Ratio** set to 1:1 (=no compression). The other type of distortion is usually caused by a fast attack and release time. If you set **Ratio** on ALL and **Release** on FAST, you'll get the fastest release time possible. Try the *Distortion* presets.

The compressor isn't fast enough

If you find that the **Attack** time isn't fast enough (doesn't squash all the transients), increase the **Lookahead** until you're satisfied.

The compressor is too fast

The **Attack** times get slower if **Ratio** is set on ALL, so in order to get a late attack set **Ratio** on ALL. If you want to increase the transients but compress the rest of the signal, use the "all buttons in" mode. Try the *Transient* presets.

It makes crackling sounds on the transients

This is because of the extremely fast attack time. First you can slow down the **Attack** times so that the gain reduction becomes not as abrupt as it was. To compensate for the slower attack time, increase the **Lookahead**.

All the energy and attack of each note disappears

It is a fast compressor and chances are that it will compress fast transients (yeah?). That's why we added the **Parallel Inject**. By increasing the **Parallel Inject** you can blend in a little bit of the original signal (with the original transients). Try to balance it so that you get the transients from the dry path and the rest of the signal from the compressed path.

All the energy and attack of each note disappears (pt. II)

Another trick you could try, if you'd like to restore some of the transients is to narrow down the detectors frequency bands by adding some **Low Cut** and **High Cut** filtering. It doesn't work on all program material, but it's worth a try.

I want some cool drum bus tricks

Ok, here are our favorites:

1. **Fat Bass Drum**: Apply heavy compression on your drum bus. Increase the **Low Cut** knob until the bass drum doesn't trigger the compressor (at about 200-300 Hz). In most cases, this will make the bass drum sound fatter and louder compared to the rest of the kit.
2. **Pumping Drums**: Set **Ratio** on ALL, **Release** on SLOW and adjust the **Input** until you're satisfied. Blend in some dry signal with **Parallel Inject.**
3. **Sustained Drums with Transients**: Apply heavy 20:1 compression with fast release and attack times, blend in some transients with **Parallel Inject**.

Adding color

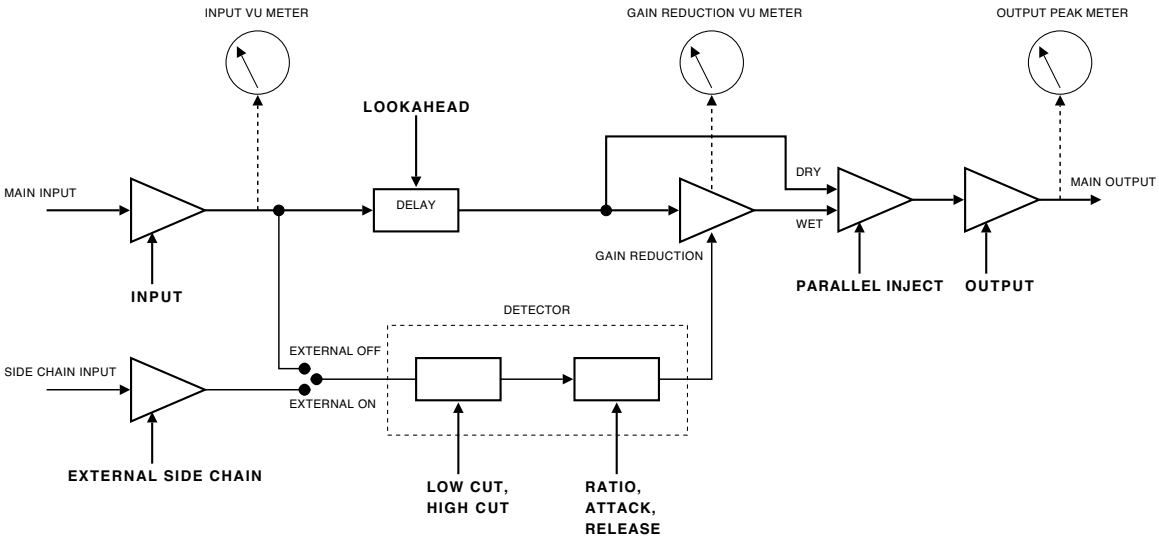
Finally you shouldn't be afraid of using this plugin in the 1:1 mode. You won't get any compression, but it will color the sound. And the meters look nice too.

Block Diagram

Below is a block diagram over the FET Compressor functionality. Please note that this illustration depicts the functionality of the plugin – the inner workings are quite different. The experienced reader will for example notice that some parts, like the feedback connection of the compressor, have been omitted in the illustration for the sake of clarity.

Credits

Oscar Öberg – modeling and implementation. **Torsten Gatu** – framework programming. **Niklas Odelholm** – framework programming and graphic design. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering and graphics. **Dan Lumbye** – A/B testing. **Stefan Fandén** – feature hunter. Thanks to **Lars Nygaard** and **Anders Bech** at Cyberfarm (DK) and **Per Åkesson** at Care Of Sound Studio for letting us use and abuse their equipment.



23 Fix Doubler

DOUBLE TRACKING OR OVERDUBBING of tracks and instruments is a studio practice that goes back to the early roots of recording. Although pioneers like **Les Paul** experimented with overdubbing on recordings during the 1930s and 1940s, this method wasn't adopted as a studio practice until the arrival of the multi-channel tape recording machines. While overdubbing became a studio standard, especially on lead vocals, time constraints and demands from musicians led to the next innovation in dubbing: *Auto Doubling*.

While some engineers already had experimented with short slap-back echo to achieve a similar effect, this technique was further developed in the 1960s with the "*Artificial Double-Tracking*" of **Abbey Road Studios**. With the rise of digital signal processing in the 1970s, these techniques were further developed and gave birth to the first digital harmonizers and doublers. Dedicated digital units were also developed by this time, but were strictly limited to studio works due to their size. Later, the Flanger effect, a side-effect of auto doubling, was popularized as an effect for guitar and drums.

This plug-in is based on the Real Time Auto Doubler AD-2 by Paul Wolff. It was a unique hardware designed to do classic tape-style doubling, and quickly became loved by musicians and sound engineers, but unfortunately never made it into serial production. The Fix Doubler is what the AD-2 was, and more. Paul Wolff has expanded the original concept and introduced several new techniques in order to give you a auto doubler that combines a classic sound with the versatility of the digital world. The Fix Doubler delivers texture and extraordinary richness on vocals and polyphonic instruments.

1. Engage
2. Stereo
3. Auto double
4. Dual sweep
5. Delay meter
6. Waveform type
   Triangle, Sine, Random
7. Mix
8. Regen
9. Delay offset
10. Sweep
11. Rate

User Interface

The Fix Doubler panel contains the following ten controls.

Mix Sets the balance between the doubled and the direct signal. Set it to INPUT to only get the input sound, or fully to DELAY to only hear the delayed signal. Set it somewhere in the middle (30-70%) to get a nice doubling effect.

Engage Use Engage as a smooth bypass if you want to turn on or off the effect during a song.

Stereo Turn on to widen the stereo image, or turn a mono signal into a stereo signal. Make sure to insert the plug-in as a STEREO or a MONO-TO-STEREO plug-in for the widening effect to work.

Dual Sweep can also be used to widen the stereo image or turn a mono signal into stereo.

Regen Sets the amount of delay feedback. Turning **Regen** up makes the effect sound fuller, but too much **Regen** will make it sound flangy and hollow. This control is not active (greyed out) when **Auto Double** is in use.

Auto Double Turns on and off the AUTO DOUBLE mode. The AUTO DOUBLE mode will put the secret sauce in the stew and make everything sound great!

Delay Offset Sets the delay offset of the doubled signal. Longer delay times will yield a more slap back kind of doubling while shorter delay times will sound more flanger-like.

Sweep Sets the balance between the fixed **Delay Offset** and the sweep. Turning this knob clockwise will cause the sweep to affect the doubled signal more.

Rate Sets the speed of the sweep, ranging from 0.01 milliseconds to 6 milliseconds. The sweep amount is also reversely affected by sweep rate (see below), so that faster rates makes the sweep less apparent. This is by design to ensure a more natural doubling sound and a constant pitch change in the delayed signal.

Dual Sweep Turn this on to get two independent doublers out of phase with each other for the LEFT and RIGHT channel. This can make the doubling effect sound fuller, and also create a nice stereo effect for mono tracks.

Sweep Type Selects the type of waveform used for the sweep: TRIANGLE, SINE and RANDOM.

TRIANGLE: Constant pitch during the sweep, suitable for low **Sweep** settings and **Dual Sweep**.

SINE: Pitch is changing throughout the whole sweep, nice on monophonic instruments and higher **Sweep** settings.

RANDOM: The pitch shift is random, and will sound more natural when used sparingly. Low **Sweep** and **Rate** is preferred for this one.

Tweaking Tips

Here are a few pointers to guide you when using the Fix Doubler:

Delay Offset

Using 20 to 45 milliseconds of delay and subtly mixing the delayed signal with the original tends to "fatten up" the sound, especially when using the STEREO or DUAL SWEEP modes. This creates an illusion of more than one singer or player and is also an excellent method of adding lushness to string or horn sections. Instruments with sharp transients, such as rhythm guitars or drums, will typically require a shorter delay, vocals medium delays, and strings and horns slightly longer delay times.

Mix

Mix ratio is essential to the perceived "fatness" of the resulting signal. A completely wet signal (**Mix** fully clock-wise) can be OK when using more extreme settings in order to create vibrato effects, but normally a mix between 20% and 50% is recommended to create a subtle, natural sounding vocal doubling or chorus.

Sweep

The Sweep is the key ingredient when creating a subtle or fat doubling chorus. Too much **Sweep**, and it will sound pitchy and wobbly, too little and you'll just get a slap back echo effect.

Chorus Effect

A natural sounding chorus effect is easily obtained by using the auto double function while setting the delay offset at a low setting (around 20–30 milliseconds), **Rate** at around 3–4 Hz and **Sweep** almost all the way to OFFSET.

Tape Wow and Flutter

By using a short **Delay Offset** at around 7 ms, **Sweep** set to 2–3%, RANDOM waveform with slow **Rate**, and **Mix** fully clockwise, you will get a tape machine like wow and flutter effect.

Original Auto Doubler hardware, the Real Time Auto Doubler AD-2 from 1979.

Vibrato

A vibrato effect can be achieved by using a completely wet signal, a fast sweep (**Rate** at 5–6 Hz), **Sweep** at 5–10%, and SINE or TRIANGLE waveforms. Use **Delay Offset** and **Regen** to get different characters of vibrato.

Credits

Paul Wolff – idea and original reference. **Arvid Rosén** – model analysis and implementation. **Niklas Odelholm** – programming and GUI. **Kristofer Ulfves** – User manual and initial testing. **Paul Shyrinskykh** – quality assurance. **Ulf Ekelöf** – 3D rendering and graphics.

24 Fix Flanger

FLANGING IS A MODULATION EFFECT that has fascinated music creators and sound engineers for decades. The instantly recognizable "swoosh" sound of a flanger has been widely used both subtly, to add weight and depth to horn sections, as well as heavily for very obvious sci-fi sweeps on everything from the sacral synths of **Jean Michel Jarre** to the big drums of **Led Zeppelin**.

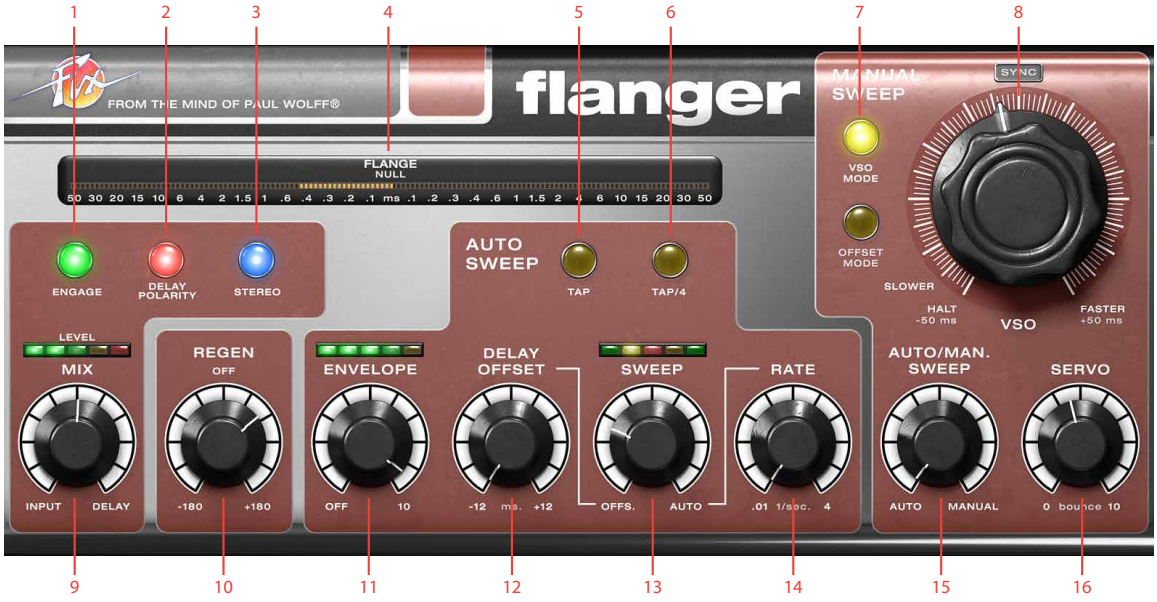
The first flanging experiments are attributed to legendary guitarist and sound engineer Les Paul in the 1940s. But it was during the recordings of the **Beatles** classic *Revolver* that the effect was achieved when **John Lennon** and studio engineer **Ken Townsend** played back the same sound through two synchronized tape machines, and lightly pressed the flange of one of the playback reels about 20ms while recording onto a third machine. This slight delay mixed with the original signal caused a flanging effect—a swirly, metallic jet type of sound.

In the 1970s, dedicated flanging machines came up but were limited to studios due to their large physical size. Later, the flanger was popularized by the flanger guitar pedals that relied heavily on analog delay with the bucket brigade echo technique.

In 1979 **Paul Wolff** formalized a vision. This vision was a flanging and doubler unit with the sound and user experience of classic tape flanging and vocal doubling, but with the advantages of electronic circuits rather than mechanical tape reels. This vision was turned into a few hardware boxes loved among musicians and sound engineers alike, but they unfortunately never made it into mass production.

The **Fix Flanger** and **Doubler** is the reincarnation of his vision, two great plug-in effects featuring the versatility of digital world combined with warmth of the analog world.

For more information regarding the Fix Doubler, please see the Fix Doubler chapter.

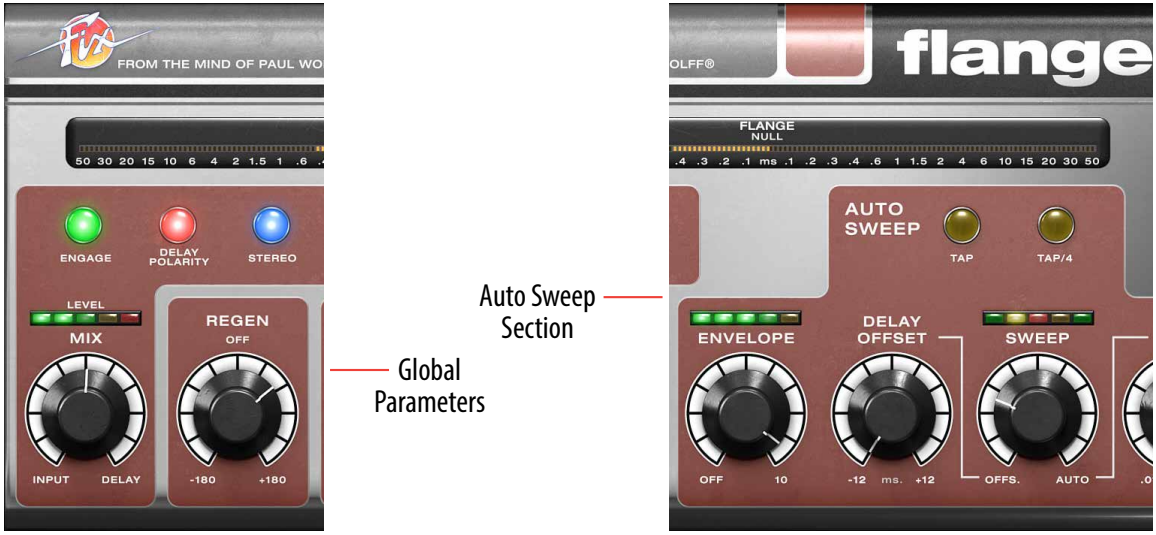


1. Engage
2. Delay polarity
3. Stereo
4. Flange meter
5. Tap
6. Tap/4
7. VSO mode, Offset mode
8. VSO
9. Mix
10. Regen
11. Envelope
12. Delay Offset
13. Sweep
14. Rate
15. Auto/Man. Sweep
16. Servo

User Interface

The Fix Flanger consists of two main sections, AUTO SWEEP and the MANUAL SWEEP. These sections both control the flanging effect, but AUTO SWEEP acts more like a regular LFO swept flanger, while MANUAL SWEEP lets you control the flange just like you would if you had two synchronized tape machines. Outside of these sections you have two controls that globally affects the amount of flanging, **Mix** and **Regen**.

Mix Sets the balance between the flanger and the direct signal. The plug-in only outputs the original signal when set to INPUT, and only the delayed signal when set to DELAY. To hear the flanging effect between the input and the delayed signal, make sure that you set **Mix** somewhere around 50%.



Regen Sets the amount of delay feedback in use. Full positive or negative regeneration will set the flanger into self oscillation. It's a nice effect, but watch your speakers!

Engage Use **Engage** as a smooth bypass if you want to turn on or off the effect during a song. The DAW's built-in bypass can cause artifacts when turning on/off the Fix Flanger.

Delay Polarity This is the polarity of the feedbacked signal. Different polarity of the feedback results in different sound of the flanging effect.

Stereo Enable **Stereo** to get a wider stereo image. If you are using a STEREO input and output, you will get a wider and more vivid image. If you are using a MONO input with a STEREO output, the **Stereo** function will make a stereo signal from your mono input. If you are using this function in mono-to-mono, the flanging will sound a bit fuller and a tad different, but won't make much sense otherwise.

Auto Sweep section

Envelope Use **Envelope** if you want the level of the input signal to affect the flange. With **Envelope** active, a loud sound will force the delay of the flange towards zero, which is a very cool effect on drums. Use sparingly to get a more random flange effect on for instance vocals.

Delay Offset Sets the starting point of the AUTO SWEEP. With DELAY OFFSET set to 0, the sweep will be both positive (delayed signal) and negative (pre-delay). If you just want a typical delayed flange, set **Delay Offset** to a positive value.

Sweep Sets the balance between the **Delay Offset** and auto sweep. When **Sweep** is set all the way counterclockwise, the **Delay Offset** determines the delay time of the flange. When set all the way clockwise, the full range of the sweep will make the delayed signal sweep back and forth +/- 15 milliseconds across the zero crossing point.

Rate Set the sweep rate, ranging from 0.01 Hz (100 seconds) to 4 Hz (¼ second).

Tap Tap repeatedly to set the sweep rate. This button cannot be automated.

Tap /4 Divide the current sweep rate by four. This button cannot be automated.

While using the envelope, the best results are often achieved by using a moderate amount of envelope modulation on the flange.

Manual Sweep section

VSO In OFFSET MODE the **VSO** knob sets the delay. In VSO MODE, it sets the speed of the delay change (see "VSO Mode" below).

Auto/Man. Sweep Sets the balance between AUTO SWEEP section and the MANUAL SWEEP section. When set fully anti-clockwise, only the AUTO section will affect the delayed signal. And of course, when set fully clockwise, only the MANUAL SWEEP section will affect the delayed signal and thus the flange effect.

Servo The amount the delay bounces when the delay is changed.When set fully clockwise, this knob will induce a form of "sluggishness" and smoothness when quickly changing the offset or simulated speed of the slaved tape (VSO). **Servo** will also affect the parameters on the AUTO SWEEP, but to a lesser degree.

VSO mode In VSO MODE, the **VSO** knob emulates how a VSO (Variable Speed Oscillator) knob works on a tape machine. By having two tape machines, where you vary the speed of one machine while blending both outputs together, you achieved the original Beatles style tape flanging. The VSO MODE emulates this behavior. This means that rather than directly setting the delayed offset between the dry and delayed signal, the **VSO** knob will

set the relative speed of the slaved tape recorder and thereby creating slightly different kind of flange behavior. When the **VSO** knob is set to -50 ms, also marked HALT, the tape reel will stop.

Offset Mode In OFFSET MODE the **VSO** can be used to directly set the delay the audio, ranging from -50 to 50 ms.

Tweaking tips

The Flange effect is highly dependent of the setting of the **Mix** knob, since the typical hollow flange is created by the differences between the delayed input and the original input.

Since the Fix Flanger can delay the signal with negative delay times, the dry signal passing through the plug-in will always have a 50 millisecond latency (look-ahead) which makes it less suitable for using live. However, Fix Flanger will always work well as post-recording treatment, and all major DAWs will automatically correct this latency.

Through-Zero Flanging

For true "through-zero flanging" set the **Mix** balance to 50%, use negative **Delay Polarity** (lamp indicator is ON) and set **Regen** to OFF. This will completely cancel out the signal when the delay passes through 0 milliseconds.

Using the VSO mode

Since VSO MODE emulates much how an old-school tape flanging rig works, one has to imagine flanging as trying to keep "the second tape-machine" in near sync by speeding up and slowing down its speed against the dry signal. This is achieved by looking at the flange meter and trying to keep the flange as much as possible in the middle by "chasing" the through-zero pass-through with the VSO knob. The servo motor emulation amount can also be handy in order to induce some natural "motor lag" when chasing the flange up and down the sonic spectrum.

By using VSO MODE with a completely wet signal (**Mix** set to DELAY), you can emulate a tape stop by turning the **VSO** knob all the way counter-clockwise.

Chorus Effect

By using the AUTO SWEEP section and setting the **Sweep** knob almost fully to OFFSET (2 – 5%), you'll get a nice chorus-like effect. Adjust the **Mix** between 25-75% to set the amount of the effect. **Delay Offset** should be around 10–15 ms, but by setting it to positive or negative offset you can get different styles of chorus.

Credits

Paul Wolff – idea and original reference. **Arvid Rosén** – model analysis and implementation. **Niklas Odelholm** – programming and GUI. **Kristofer Ulfves** – User manual and initial testing. **Ulf Ekelöf** – 3D rendering and graphics. **Paul Shyrinskykh** – quality assurance.

25 Fix Phaser

The Phaser was originally invented by a sound engineer in the mid 1960s while trying electronically emulate the effect of a rotating Leslie-cabinet. Although similar effects had already been produced by others with experimental equipment earlier, the first commercial product to feature the phasing effect did not surface until the late 60s.

A phaser effect is created by splitting input audio into two different paths and then mixing them together. One path treats the signal with all-pass filters, which preserves the amplitude of the original signal but alters the phase. The amount of phase change is frequency dependent. When this phase-altered signal is mixed together with the other path, the dry signal, the frequencies that

are out of phase will cancel each other out which creates the characteristic audio phase-notches. By changing the dry/wet mix the depth of these notches are altered. The deepest notches occur when the dry/wet mix ratio is 50% (this is the preset value in Fix Phaser).

The Difference between Flanger and Phaser

Historically, there has always been much confusion about what is phasing and what is flanging: The difference between phasing and flanging is that the latter creates an unlimited series of equally spaced notches and peaks, where phaser creates a limited number of notches with uneven spacing. This means that flanging is very much a similar sounding effect to phaser, but in the Flanger, the notches are linearly spaced in the frequency domain. Flanger is typically created by mixing a delayed version with itself and often tend to sound more pronounced and natural ("jetplane"), whereas phaser often sound more electronic and otherworldly ("space-whosh").

Overview

The Fix Phaser is a very versatile effect with the ability to re-create everything from the phasing sounds of various classic gear to the more vintage sounding vibratos derived from the late 60s. Like the other Softube Fix products it has a dry/wet Mix knob giving the user the freedom to freely set their desired blend of dry input and the fully wet signal.

The Fix Phaser features three main phasing modes – Normal, Effect and Wide. Normal mode is the classic phase-shifting sound with all the liquid transparency, whooshing sweeps and tingling swirls we're all used to. Effect mode adds a pronounced chasing stereo effect where the left channel's sweep is out of phase with the right channel's sweep, and Wide mode further expands on this principle where a subtle widening effect is still apparent even when using small sweep amount.

The Fix Phaser is swept up and down by a sweep-generator with variable amount, rate and sweep type.

Finally, the Fix Phaser also feature an envelope-follower with fast and slow mode, positive and negative response, that makes it possible to use create auto-wah style effects among other things.

User Interface

The Fix Phaser panel contains the following twelve simple and easy to use controls. This is a brief description of each and everyone:

Engage (Out/In)
This button turns effect on and off. Effect is bypassed when Engage is set to "out".

DAW Sync (Sweep/Sync)
This locks the rate of the Sweep generator to the tempo of your DAW when set to "Sync" (DAW sync lamp is lit).

Stereo (Normal/Effect/Wide)

- **Normal** – this is classic phaser mode where both sides of the stereo input is treated equally.
- **Effect** – enhances the stereo effect of the phaser by using the same phase-shift on left and right side but with different phase on the sweep. This gives the phaser more of a "chaser"-kind of effect where one side always is chasing the other.
- **Wide** – left and right phase will change in amplitude and polarity in a preset ratio to the sweep

1. Engage
2. Daw Sync
3. Stereo (Normal/Effect/Wide)
4. Slow
5. Modal centers
6. Stages (2, 4, 6, 8, 10)
7. Sweep type (Triangle/Sine/Eliptic)
8. Mix
9. Level meter
10. Regen
11. Envelope
12. Phase offset
13. Sweep
14. Sweep meter
15. Rate

in addition to the effect mode. Wide mode makes the soundstage wider and creates a subtle, musical movement which is very handy when mixing static program material.

Note: When using Phaser with Stereo mode set to Wide, the sweep modulation will always affect some aspects of the phasing with a preset value which is not affected by the Sweep (amount) knob. This is by design and intentional behaviour.

Stages (2,4,6,8,10)
These buttons set the amount of phaser stages used. Two stages creates one notch, 4 stages creates two phaser-notches etc. This is one of the key elements to look at when re-creating the phaser sound of classic gear.

Sweep Type (Triangle/Sine/Elliptic)
Selects the type of waveform used for the sweep:

- **TRIANGLE-waveform:** Suitable for creating a regular, ongoing phaser movement.
- **SINE-waveform:** Can be used for more natural sounding phaser movement.
- **ELLIPTIC-waveform:** This sweep type spends more time on high frequencies than lower ones.

Slow (Fast/Slow)
This switch determines how swift the envelope follower will react to an incoming signal. Use Fast response for drums, rhythmic guitar or clavinet picking, and Slow response for sustained chords and synth-pads.

Mix (Dry/Wet)
This knob set the mix between dry (uneffected) input signal and the fully wet signal. When set to fully wet the Fix Phaser can be used as a vibrato-like effect, bending input notes up and down in sync with the sweep rate.

Regen (0-100%)
Sets the amount of feedback for the end of all pass chain. A large amount of Regen make the Fix Phaser sound more electronic and out of this world.

Envelope (-100% - 0 - 100%)
This is the amount of which the envelope follower will affect the modal center movement. It is a polarized knob which means that the counter-clockwise position at -100% means that it will have a negative sweep impact on the phaser movement. AWith the knob at 12 o'clock it has no effect (default value), and at fully clockwise, it will have full positive effect on the phaser movement.

Phase offset (20hz - 20kHz)
Sets the offset of the phased signal.

Sweep (0-100%)
This knob sets the Sweep amount to affect the phaser movement back and forth.

Rate (0.01hz - 6hz)
Sets the speed of the Sweep, ranging from 0.01 to 6 hz. This means that, when Rate is set to lowest the lowest speed, 0.01 hz, it will take 100 seconds for the sweep generator to complete its cycle, while at the highest rate completion will occur 6 times per second.

When DAW sync is set to “Sync”, Rate is divided into 8 bars, 6 bars, 4, 3, 2, 1.5, 1 bar, half bar, dotted quarter, quarter and eighths parts of a bar. These values represent how many time-measures it takes for a sweep cycle to complete.

Meters

Modal Centers
This meter on the upper part of the plugin shows how many phase notches that are active and approximately on which frequency they are currently at work. More stages selected by the stage buttons means more notches.

The modal center display is also a handy tool in order to understand the envelope sweep and polarity, phase offset and sweep rate and range.

Level
This meter reflects output level. Keep this meter in the green or yellow for optimum response. Lower input volume or Regen if this meter is in the red.

Sweep
This meter shows the movement of the sweep generator at all times and is not affected by the Sweep (amount) knob.

Tweaking tips

Here's a few pointers to guide you when using the Fix Phaser:

Mix ratio is essential for getting the most effective phase-cancellation - the preset value 50% is will give you the deepest phase-cancellation. If the mix ratio is raised towards 100% wet, the phase-effect will get more "pitchy", especially with many stages selected.

This "pitchy" effect can be used to create vibrato effects with the Fix Phaser. Set Mix ratio to 100%, use 6 or more stages and a fairly high (30% +) sweep and you'll start hearing pitching effects.

By using extreme mix settings (100% wet) in combination with the Stereo "Wide" mode, tremolo, as well as combined "tremolo + vibrato" effects can be achieved in the same way.

The number of stages selected will greatly affect your phaser sound. Try for example a slowly sweeping phaser using 2 or 4 stages with high Regen. Now, try instead 10 stages and listen to the difference. The more stages, the more "hollow" and "electric" phaser characteristics.

You can also use the Phaser with a low dry/wet ratio (25-30%), low sweep (30%) with a fairly swift sweep rate (3 hz), and no regen in order to create a "Chorus"-like effect.

The envelope-follower feature of the Fix Phaser is great for creating auto-wah like phasing. Use the Fast mode for staccato, funky fast playing or drums, and use the Slow Mode for sustained chords, strings and pads.

Finally, the Fix Phaser can also be used as a subtle or not so subtle stereo widener in the Stereo "Wide" mode. Using the Phaser with 60-70% wet mix, many stages selected and Sweep set at minimum (0%), the Regen and Offset knobs can be used to set the focus and "hollowness" of the stereo movement. The speed of this movement is set by the Sweep Rate knob.

Credits

Paul Wolff – idea, testing and feedback. **Arvid Rosén** – model analysis and implementation. **Paul Shyrinskykh** – algorithms testing and feedback. **Niklas Odelholm** – programming and GUI. **Kristofer Ulfves** – User manual and testing. **Joe Chiccarelli** - Presets and testing. **Igor Miná** - User Manual layout. **Ulf Ekelöf** – 3D rendering and graphics.

26 Focusing Equalizer

THERE ARE MANY EQUALIZERS OUT THERE. Some are extremely advanced with built-in spectrum analyzers and intelligent super modes. Others are simple and nothing more than your average digital equalizer. While some claim to possess super powers, the differences are usually just in the user interfaces, and not always in the sound. Others claim that there is only one way to make a digital equalizer, and therefore every digital equalizer sounds the same. That's not true. Modeling a three band parametric equalizer is more than taking the three separate bands and putting them together. The real electronics interact with each other so that the sum becomes more than its parts, and this sum isn't possible to re-create by stacking up a bunch of standard digital equalizers.

We set out to model two vintage equalizers as exactly as possible. The result are the PASSIVE and ACTIVE equalizer that are probably installed on your computer right now. The ACTIVE is the most traditional of the two. Use it as a standard equalizer and feel comfortable about it sounding the way the real gear does. The PASSIVE is a bit more quirky but just fantastic sounding. An excellent help when you need that glimmering high-end or warm low bottom or *je ne sais quoi*...

If you combine these two, you'll get what we call the FOCUSING EQUALIZER. It is designed to be a fast and efficient work horse that immediately produces the desired results. The combination of three bands that automatically follows the High and Low cuts and a compressor-saturator-distortion-mojomaker-tool makes it great tool to tidy up and fix your tracks.

We hope that you'll enjoy this set of equalizers and that you will find that they are excellent additions to your toolkit that will help you make great sounding music!

The Trio ("Passive-Active Pack")

The FOCUSING EQUALIZER is a part of a trio together with the PASSIVE EQUALIZER and the ACTIVE EQUALIZER. The PASSIVE and ACTIVE have very different sounds, while this unit combines these two into a new type of equalizer with a classic sound. We can assure you that there are no other equalizer plug-ins that sound like these.

At a first glance, you will notice that these three units don't look like your average equalizer, and at a closer inspection you'll see that the knobs behave in a peculiar way. This is because we have chosen to keep the way the original units worked, and while it feels quirky in the beginning you will get used to it pretty fast.

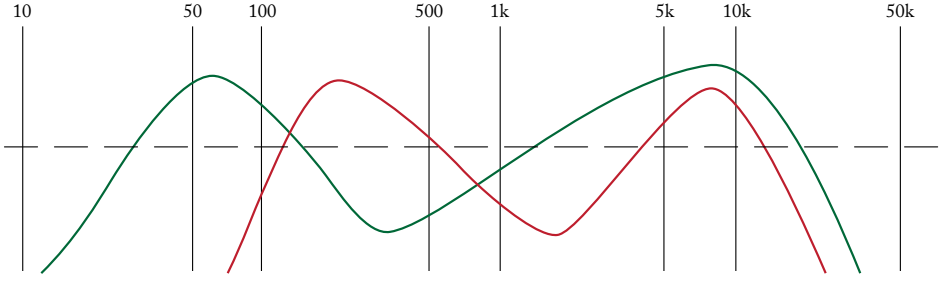
The Story

The FOCUSING EQUALIZER was actually the main goal of this project. We have had the idea for a long time, and we developed prototype after prototype to try the concept. It sounded good, but we had a hard time getting the right character. While showing it to one of our main dealers in Stockholm he asked us what kind of filters we used and why we didn't model a great sounding passive equalizer and put that inside the FOCUSING EQUALIZER instead of the off-the-shelf EQ algorithm we used in the prototype. At the store he had tons of equalizers and we listened to almost all of them, not really satisfied with what we heard. At the end of the visit he brings this Neumann and says, "Oh, I almost forgot this one. We had it here for some reparations, but the customer won't notice if you borrow it for a while!" We fell instantly in love with the sound, and when we put it inside the FOCUSING EQUALIZER we immediately got what we had been looking for. One drawback with the Neumann is that it isn't very sharp, and sometimes difficult to use to really control and shape the sound. One of the runner-up equalizers we tried was the Filtek, but we didn't really want to model it because of the extreme complexity. Eventually (and with some persuasion from our testers) we realized that the Filtek was exactly the counterpart to the Neumann that was needed for the FOCUSING EQUALIZER and the PASSIVE-ACTIVE PACK.

What Is Unique About the Focusing Equalizer?

The combination of three bands that automatically follows the high and low cuts. So if you set your low cut at 2 kHz, the low band will automatically get a center frequency at or above 2 kHz. The mid band will have its center frequency somewhere between the high and low cut. Sounds easy enough?

The graph below shows two settings, using the ACTIVE filter type. **Low** and **High Bands** are on full boost, and **Mid Band** on full cut. The difference between red and green graph is the settings of the **Low** and **High Cuts**, 50 Hz-12 kHz (green) and 200 Hz-10kHz (red), respectively. As you can see, the mid dip follows the settings of the cuts.



Work Flow at a Glance

1. Set the frequency range with **High** and **Low Cut.** This will **automatically** change the frequencies of the **Low**, **Mid** and **High** bands).
2. Choose **Equalizer Type** (tip: start with ACTIVE)
3. Adjust the equalization (**Low, Mid**, and **High Gain**)
4. Toggle **Equalizer Type** to hear how the sound changes
5. Add some **Saturation** if needed.

Work Flow

First of all, you should disregard the **Saturation** controls and only focus on the **Low** and **High Cut** faders. Try to set these so that you filter out all the unwanted parts of the signal. For most instruments other than bass and bass drum, you can go pretty high up with the **Low Cut** without really changing the sound. By setting the **Equalizer Type** to PASSIVE you will get a smooth and musical cut, while on the ACTIVE setting you get a more brutal filtering, more like a standard parametric equalizer.

Step 1: Frequency Range Adjust (Low and High Cut) and Equalization

After adjusting the **High** and **Low Cut** faders, the next step is to use the **Low**, **Mid**, and **High Gain** to shape the signal. These equalizer bands have been automatically set to natural frequencies within the selected frequency band. In practice, this means that the **Low**, **Mid** and **High** will always make sense.

> If the **Low Cut** is set at 1 kHz and **High Cut** at 5 kHz, the center frequencies of the **Low**, **Mid** and **High** will always be within the 1 and 5 kHz range.

Step 2: Set Equalizer Type

When you have set the right character of your track, you can toggle between the ACTIVE and PASSIVE **Equalizer Type**, to see what character you prefer. In general, the ACTIVE is most suitable for narrow, focused sounds, while the PASSIVE is more suitable for wider sounds. The PASSIVE has (in our ears) an acoustic feel with a typical 60's glow, while the ACTIVE typically gives you a controlled 70's hard hitting sound.

> Try to avoid setting the **Low** and **High Gain** on full blast while using the Passive equalizer type. By doing this, you will effectively cancel any effect from the **Mid Gain**.

Step 3: Use the Saturation and Saturation Type

The **Saturation** knob is a very effective way of boosting the energy of the signal. You can use the small saturation meter in the top-left corner of the frequency range window to see how hard it is working. It is not necessary to use the saturation at all, but it is a very useful feature.

1. Saturation
2. Low cut
3. High cut
4. Output volume
5. Saturation type
6. Low gain
7. Mid gain
8. High gain
9. Equalizer type

Knobs

Saturation Sets the amount of saturation. The saturation circuit is completely bypassed if the knob is set in the minimum position.

Saturation Type With KEEP LOW you will be able to distort the track without getting a farty bass sound. KEEP HIGH will do the same, but for high frequency material. Neutral will not do anything special with either the high or low frequencies.

Low and High Cut 6dB/octave (PASSIVE mode) or 18dB/octave filters (ACTIVE mode).

Low, Mid and High Adjust the gain of the filters. The **Low** and **High Gain** will only boost the low and high frequencies. If you need to cut, you will have to use the Low and **High Cut** faders.

Equalizer Type Choose between the PASSIVE type or an ACTIVE type equalizer. Affects all equalizer controls, but not the saturation circuit. You can also bypass the entire EQ circuit (and just use the saturation circuit) by setting this in the off position.

Output Output volume. Goes from nothing to +12 dB. At the 12 o'clock position, it's set at 0dB, and going clock wise will increase the output volume by +2 dB for each mark.

Meters

The Focusing Equalizers have the following meters and status indicators.

Saturation Meter This meter lights up whenever the saturation circuit is active and is adding saturation to the signal. Use this more as a visual indication than an absolute measurement of the amount of saturation.

Frequency Range Guides These "tuner looking" guides help you see your selected frequency range.

Stereo Indicator Lights up if the plug-in has been activated in stereo mode.

Remote Control Indicator (Pro Tools only) Whenever automation is used, or if and external control surface is connected, the Remote status indicator will light up and the (remotely controlled or automated) knobs will get a thin colored frame.

Output Peak Meter The output peak meter is a fast peak meter with hold values. If you use the plug-in in mono mode the left and right meter will show the same mono signal on both meters.

1. Saturation meter
2. Remote control indicator
3. Stereo indicator
4. Output Peak meter
5. Frequency range guides

Differences Between Passive and Active

This table highlights some of the differences between the three equalizers. Please note that there are a lot of other differences that make up for the characteristic sounds, and not many of these are easy to put in a table like this.

1) The Passive EQ doesn't have any cutting bell filters.

2) "Constant Q" wasn't defined at the time that the Filtek was built, but it shows many properties of a "Constant Q" equalizer.

3) The cut filters are shelving in the Passive Equalizer (due to parasitic resistance in the inductor) while this (unwanted?) property have been removed in the Focusing Equalizer.

Finally, a Note on Modeling

There are lots of buzzwords when it comes to recreating old gear in the digital domain. Many are those who claim they "model" stuff, few are those who actually do it. The advantage of real, component-based, physical/mathematical modeling is that you capture the entire behavior of a unit in one big sweep. The disadvantage is that the math can get very complicated, even for circuits that doesn't look complicated at all. Even a simple thing like a bypass (such as in the Filtek) is complicated and causes problems when modeled, and it would be easy for us to take a short cut and mimic stuff when the modeling gets tough. But we don't do that at Softube, because we believe that you can hear the difference. So we do real modeling, for good and for worse.

Credits

Niklas Odelholm – modeling and graphics design. **Torsten Gatu** – concept. **Oscar Öberg** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – graphics rendering. Thanks to **Stefan Fandén** and the crew at Deluxe Music for letting us borrow the gear!

	Active Equalizer	Passive Equalizer	Focusing Equalizer
Symmetrical Bell Shapes	Yes	N/A (1)	Symmetrical (ACTIVE), Non-symmetrical (PASSIVE)
Constant Q	Yes, sort of (2)	No	
High/Low Cut	18 dB/oct.	Shelving, 6 dB/oct. (3)	6 dB (PASSIVE), 18 dB (ACTIVE)
Filter Types	Bell 3-pole filters	Shelving 6 dB/oct. (High/Low) Bell 1-pole filter (Presence)	
Isolated Bands	Yes	No	

27 Harmonics

After more than a decade of analyzing and recreating the sound of classic studio hardware, we've hand-picked the best-sounding components from legendary units and placed them in one box to deliver a musical and versatile palette of analog flavors.

User interface

1. THD/Level meter
2. Drive types
3. Meters
4. Dry/wet
5. Pre/post distortion
6. Drynamics
7. Dynamics on/off
8. Drive
9. Character
10. High cut
11. Low cut
12. Filter type
13. Input
14. Input level
15. Output
16. Output level

Parameters

Drive types

Solid Inspired by units with hand-made quality solid state components from the early 70s. The distortion type will help any track to stand out in a mix, adding definition and separation in the soundstage. It will work great on vocals, drums and just about anything else.

Transf Inspired by the transformer sound of American consoles. The transformer will add the signature low-end tone of countless records from the analog mixing era. Full mixes, kick and bass tracks will shine when using this distortion type.

Master Distortion designed specifically for mastering/bus applications, leaving the transients intact. This type can be subtle, adding just the right amount of analog touch to the master bus, or, if driven hard, a smooth saturation bringing character and attitude to any program material.

Tube Inspired by a boutique tube circuit found in various pieces of iconic vintage gear from the mid-60s. Delivers the unparalleled warm tube sound that will benefit any bass or guitar tone.

Modern Inspired by an English all-valve modern distortion box with triode setting. This type adds instant warmth and punch.

Other parameters

Character Use this knob to adjust the character of distortion using mastering-grade filters. Turning clockwise favours high frequencies before the distortion, while turning counterclockwise favours low frequencies before distortion. The character filters are tailored for each distortion type.

Drive The Drive knob controls the amount of distortion.

Dynamics Allows you to add dynamics back to the processed/distorted material. It uses a unique algorithm that analyses parts of the incoming audio material before distortion and reconstructs the dynamics of the original waveform. Turn the knob clockwise to steadily increase the amount of dynamics. Turning the knob counterclockwise will add thickness and energy to the audio material, and bring up all the lower details. In combination with Drive it will add an unmistakable rich tone to drums, bass and guitars.

Dynamics On/Off Flip this switch to turn the Dynamics processing on and off.

Meter Use this switch to change the VU meters between input level display and THD (Total Harmonic Distortion).

Pre/Post The high and low cut filters can be applied before or after distortion. Use this switch to change their position in the signal chain.

High Cut Filter Lower the cutoff frequency of the High Cut filter to get soft, warm tone even when a lot of distortion is applied.

High Cut Filter Type Choose between 1, 2, or 3-pole filters to define the slope and sound of the High Cut filter parameter.

Low Cut Filter Raise the cutoff frequency of the Low Cut filter to roll off sub frequencies that can trigger excessive distortion in some program material.

Low Cut Filter Type Choose between 1, 2, or 3-pole filters to define the slope and sound of the Low Cut filter parameter.

Dry/Wet Blend the Dry and Wet signals with the Dry/Wet mix knob.

Input The Input fader controls the input gain.

Output The Output fader controls the output level of the plug-in.

Meters

VU The plug-in's main VU meters can display either input level, or THD. When in THD mode, the meters show the distortion amount. It is handy switching to this mode to track even the most subtle distortion levels.

Input/Output level These meters show the level of your signal at input and output.

In use

Bass

Every Drive Type will add different and unique flavours to the bass track. One of our favourites for bass and guitar is the Tube Drive Type. Set Drive to max, add a bit of Dynamics, turning the knob clockwise, and your bass track will shine with unique colors and life like never before. Add more definition to the sound by turning the Character knob clockwise. Add more rock solid low end by turning the Character counterclockwise and blending some Dry signal back by using the Dry/Wet control. You might want to add more Input Gain for even more distortion.

You can use several instances of Harmonics in series to add different character to top and low end.

Master bus

Choose the Master Drive Type and switch the THD/Level meter to the THD position. This will allow you to track the most subtle distortion. Please note: Drive in Master type doesn't only increase the harmonics amount, but changes the harmonic content as well, experiment with the position of the knob to find the sweet spot for your track. You may alter the flavour of the the Drive even more by turning the Character clockwise, adding mastering-grade sheen and clarity to the program material. Turning the Character counterclockwise will add punch and tightness in the low end. To adjust to the changes you might use the Low Cut and High Cut filters. It is recommended to use the 3-pole Filter Type on the Low Cut and the 1-pole filter on the High Cut to get the best results when working on your master bus.

Vocals

To add an analog touch and that extra clarity to vocals, choose a Drive Type to your taste and increase the drive until you hear a fair amount of distortion. Turn the Character clockwise to add more harmonics to the high frequencies. Use Dry/Wet to blend the Dry signal back and mix to taste.

Drum loops

To add some fatness and weight to the drum loops, turn the Dynamics knob fully counterclockwise and add Drive until the desired amount of distortion is achieved. You might want to blend some of the dry signal back by using Dry/Wet knob.

Credits

Paul Shyrinskykh – product management and sound design, **Kim Larsson** – modeling, **Niklas Odelholm** – graphic design, **Ulf Ekelöf** – 3D rendering, **Maxus Widarsson** – quality assurance, **Igor Miná** – user manual layout.

MODULAR EXCLUSIVE

28 Heartbeat

SYNTHESIZED SOUNDS AND imaginary worlds have inspired musicians since the mid 1900s when Dr. Bob Moog invented the first ever voltage controlled synthesizer modules, and eventually launched the electronic synthesizer as a new instrument into the limelight of every day musicians. Around the same time, electronic organ-makers looked into ways of electronically reproducing drums and rhythmic sounds. In the 70s, the electronic drum machine made its way into the public mind and electronic drum production could soon be heard in everything from disco, electro and hip-hop to pop and rock.

This legacy of finding new and exiting electronic percussive sounds is something we want to convey in Softube Heartbeat—the joy and excitement of exploring new and interesting percussive worlds by looking back at history, but at the same time adding something new to the concept.

Introduction

Heartbeat is an innovative software drum synth with a familiar, yet unique, sound character. A world class effects section is included, as well as the innovative Auto Layer Machine which will take your beats to unexpected places. While Heartbeat draws inspiration from the best analog drum synths from the 1980s, it does not emulate any existing drum machine. The sound mostly originates from Softube's own modeled analog synthesis, which has been augmented with carefully selected waveforms.

The core of Heartbeat consists of the eight instruments. You will find two different bass drum instruments, which can be as punchy and deep as you want them, but are also perfectly capable of producing snappy and hard hitting woody textures. The two dedicated snare drum instruments have six parameters each which allows you to achieve anything from edgy rimshots, soft and whispery snare rolls to machine-like claps.

The two percussion instruments are identical and can be used to model anything from 80s style synthetic toms to cowbells and noise drops. And just like the other instruments, the hihat and cymbal channels offer flexible synthesis engines—tweak to your heart's desire! But the idea behind Heartbeat is to make it a one-stop shop for your beat creation, so we also added an effects section and the innovative Auto Layer Machine.

1. Drum channels
2. Master bus
3. Auto layer machine
4. Valley People Dyna-mite
5. Filter echo
6. TSAR-1D Reverb
7. Utility
8. Instruments
9. Mixer
10. Global

Heartbeat's Sections

The left half of Heartbeat's graphical interface is taken up by the eight **Drum Channels**. These all consist of (from top to bottom) the **Utility** section, the **Instrument** and the **Mixer Channel**. By default, the **Mixer Channels**' outputs are summed and sent to the included **Valley People Dyna-mite** compressor/limiter/gate (read more below), and then on to the **Master Channel** on the right side of the interface.

Below the **Valley People Dyna-mite** unit are the **Filter Echo** and the **TSAR-1D Reverb** effects. Each mixer channel has send knobs (labeled ECHO and REV respectively), that determine the level of sound that is being fed from the respective channels into the two send effects. The output of the send effects is then summed with the output of each **Drum Channel**, and fed into the **Master Bus**.

Above the **Valley People Dyna-mite** unit you will find the **Auto Layer Machine**. This is a device that can be used to layer sounds or trigger a chain of events, in order to create new sound textures or create automatic fill patterns in up to four steps. By pulling the **Chaos** slider to the right, an element of randomness is introduced—so Heartbeat has a mind of its own and might give you some unexpected results.

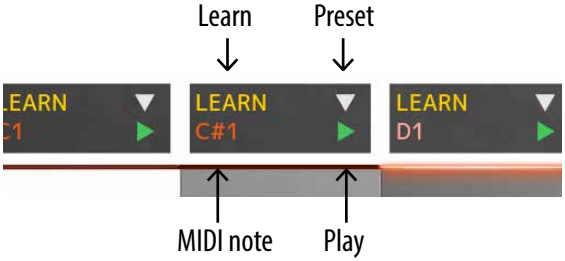
Getting Started

If you're familiar with working with software instruments, this section may be all you need to get started. Refer to the in-depth parts of this manual to learn the details.

After you have finished installing Heartbeat, open a new song in your DAW, and launch Heartbeat which you will find in the DAW's software instruments folder.

Setting up MIDI

Heartbeat is by default set up so the eight **Instruments** respond to the MIDI notes that are most commonly used for drum machines and drum software. The red text in the **Utility** section (the black square at the top of each **Drum Channel**) displays the MIDI note set up for each **Instrument**. If you would like to change it, click and hold the red text, and pull up or down. Or click **Learn** and strike the desired key/pad on your MIDI controller to assign this key/pad to the **Instrument**. Please note that the **Hihats** instrument receives input from two different MIDI channels, as it can be used for both closed and open hihat sounds. If you don't have a MIDI keyboard or pad controller available, you can use your mouse to click the green arrow in the **Utility** section, which will trigger the sound.



Presets

Clicking the white arrow will open a list of presets for that specific **Instrument** or effect. There are also presets available for entire **Heartbeat** kits (with settings for all eight instruments, effects, levels and master settings) via the usual preset function in your DAW.

Instrument and mixer

In the **Instruments** sections, you will see the settings for the instruments. These are all clearly labeled and adjusting them will yield apparent changes to the sound. Below these are the **Mixer Channels** with effects sends, a one-knob EQ (adapted for each **Instrument**) and an auto-pan function called **Ping/Pong**. The **Pan** knob and volume fader acts as you would expect, as do the **Solo** and **Mute** buttons.

Effects

The parameters of the three effects are clearly labeled. The **Pre** button to the right of **Filter Echo** and **TSAR-1D** inserts these effects before the **Valley People Dyna-mite**, which means the reverb and echo tails are also affected by **Dyna-mite's** processing.

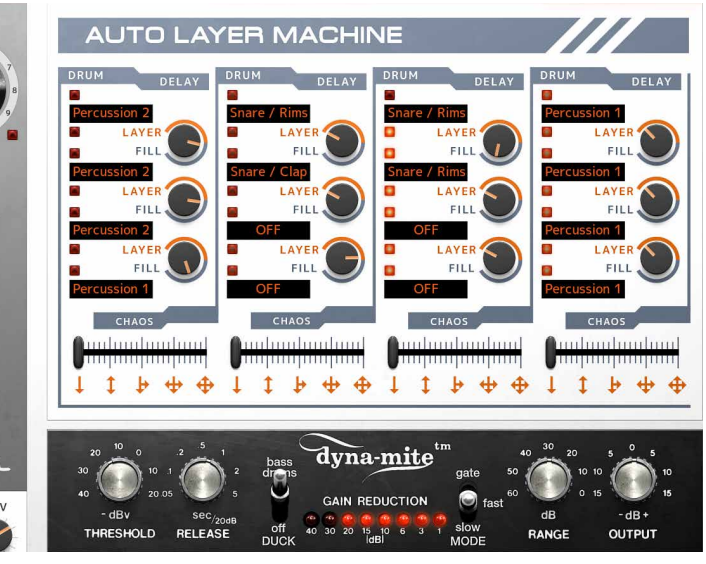
Master bus

The output from the **Instruments** and the effects are all summed in the **Master Bus**. **Mono Cut** collapses any stereo sounds below the selected cutoff frequency into mono, to ensure phase compatibility in the important lower frequencies. **Width** makes the entire stereo image wider or narrower.

Global parameters

Global parameters

At the bottom of the graphical user interface, you can determine Heartbeat's overall sensitivity to MIDI velocity, as well as separately determine how much velocity will affect **Pitch**, **Attack** and **Decay**. **Time Gate** can shorten all **Instrument** sounds independently of their velocity to create a stuttery, machine-like sound—very useful for creating variation to the sound by quickly adjusting a single knob.



Auto Layer Machine

Auto Layer Machine can be used to easily layer sounds from two or more **Instruments** for new textures, or to trigger flams or autofills. A quick way of learning what it does is to try the different factory presets and note the differences to their settings. Click the white arrow in the **Utility** section of one of the four **Auto Layer Machine** channels, and play the pattern by clicking the green arrow, or hitting the MIDI key assigned to that **Auto Layer Machine** channel (as indicated in red text in the **Utility** section). Refer to the detailed section for further information.

Keyboard shortcuts

All knobs in can be reverted back to its preset-settings by ALT-clicking on the parameter.

Fine-adjust any parameter in Heartbeat by CTRL-clicking (Windows) / CMD-clicking (Mac OS).

By clicking the **Setup** button below the Heartbeat logo, you can choose some basic settings for Heartbeat, such as turning off the tool tip pop-up windows.

Sound Architecture

See the image below for a description of the signal flow. The incoming MIDI signals can trigger either the instruments or the Auto Layer Machine. If an Auto Layer Machine channel is triggered, this in turn triggers the instruments.

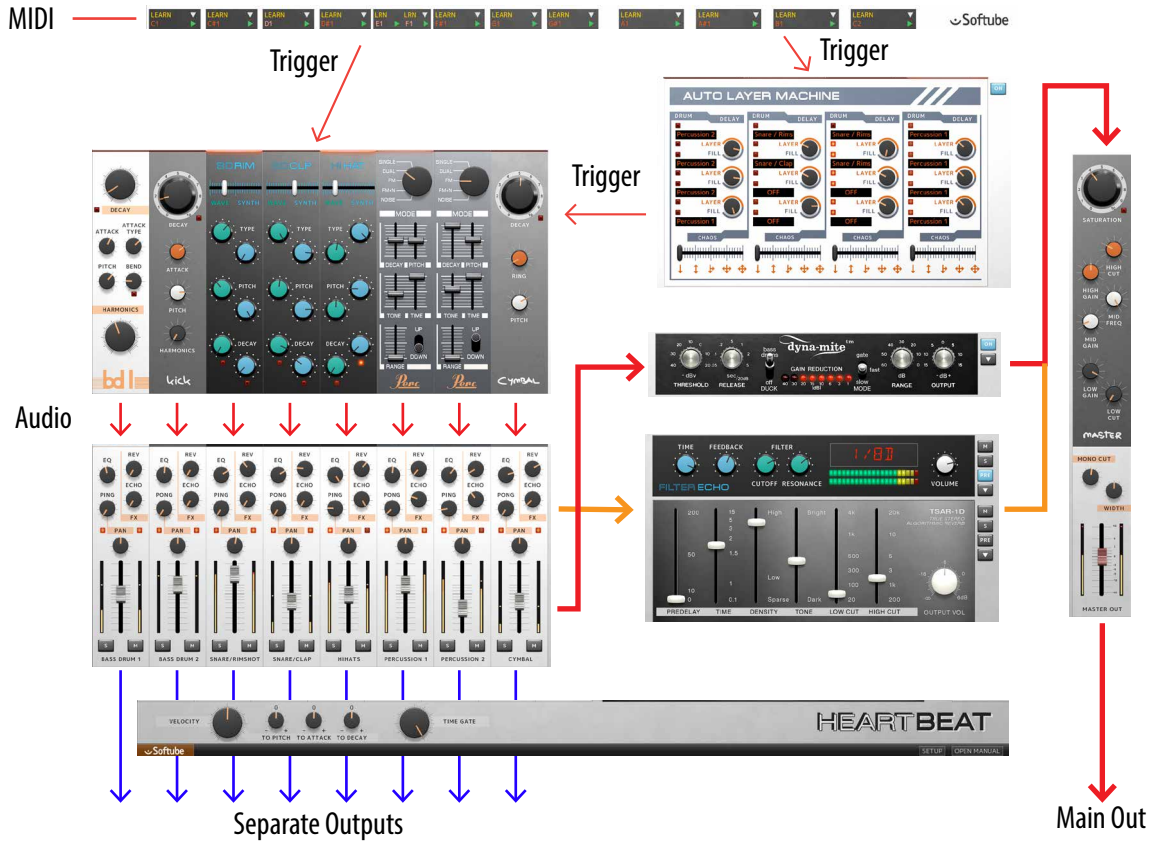
After the trigger, the instruments generate drum sounds that is routed to the corresponding mixer channels, and then routed through different paths:

1. The main signal (red) is sent through the mixer channel's volume fader and mixed with the other instruments, sent to the Valley People Dyna-mite, gets summed with the signal from the send effects, passes through the Master Bus and is eventually sent out via Heartbeat's Main Out to the DAW channel.
2. If the user chooses to, one signal is sent via the mixer channel's **Rev** send to the TSAR-1D Reverb and another is sent via the mixer channel's **Echo** send to the Filter Echo (orange).
3. One signal (blue) is sent *pre-fader* to the respective instruments' Separate Output, to be used as an isolated signal in your DAW, if this is supported by your DAW.

If the TSAR-1 Reverb or Filter Echo's **Pre** buttons are activated, the output from the effects are instead routed to the Dyna-mite, instead of directly to the Master Bus section

If **Duck** is set to BASS DRUMS, Bass Drums 1 and 2 are also routed to the sidechain of Dyna-mite, where it controls Dyna-mite's behavior.

For a more detailed overview, please see "Block Diagram" on page 234.



Utility Section

The Utility section is the black field on top of each Drum Channel (and Auto Layer Machine channel).

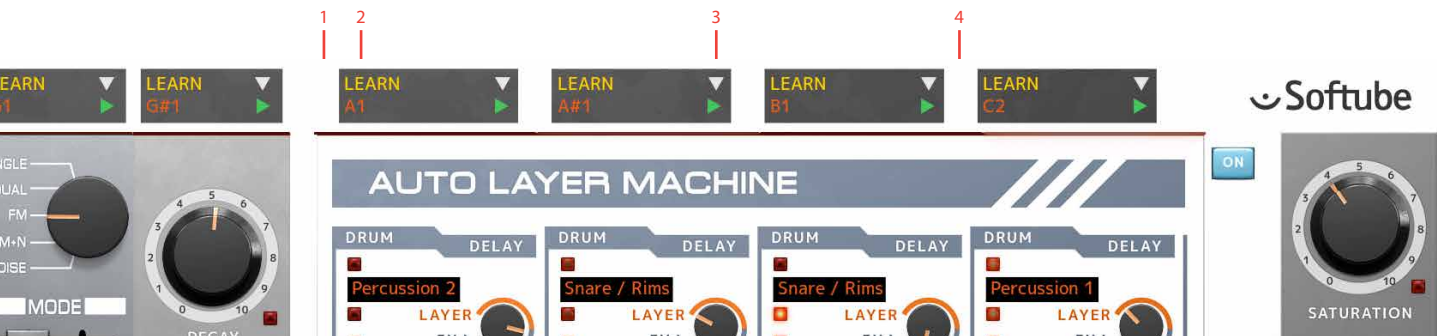
Learn The Learn function is a quick way to assign a key on your MIDI keyboard or pad controller so it triggers the corresponding Instrument, in case you would like to change it from the factory settings. Click Learn, which will start blinking to indicate that it is awaiting an incoming MIDI note. Press the MIDI key on your keyboard controller (or strike the pad on your MIDI pad controller) that you want to assign to the Instrument. The MIDI Note indicator (red text below the Learn button) will show the new MIDI note you assigned to the Instrument, and the Instrument will now respond to incoming MIDI data on that note number. Please note that the Hihats instrument has two Learn buttons, as it can be used for both closed and open hihat sounds.

MIDI Note The red MIDI Note indicator is located just below the Learn button, and tells you which MIDI note is assigned to the corresponding Instrument. You can change this by clicking, holding and dragging up/down the MIDI note number, as an alternative to using the Learn function explained above.

Channel Presets (white arrow) Click the white arrow to open the channel presets pop-up menu. This reveals a small selection of presets for each individual Instrument, intended as starting points for your own sound creation. Since both percussion channels use the same sound architecture, they also share the same channel presets. The same goes for the Auto Layer Machine channels. Only the Instrument parameters and the Equalizer (EQ) are affected by the channel presets. The effect sends (Rev, Echo), Ping/Pong, Pan, Volume, Mute and Solo are unaffected.

Play (green arrow) Clicking the green arrow will trigger the corresponding Instrument with maximum velocity. This function is handy when auditioning Instrument sounds without a MIDI keyboard or pad controller connected to the computer.

1. MIDI note
2. Learn
3. Channel preset
4. Play



The Instruments

The eight drum instruments occupy most of the upper left part of Heartbeat's graphical user interface. From left to right, you will find two different bass drum channels, two snare drum channels of which one is more suitable for typical snare sounds and the other leans towards clap sounds, a hihats channel (with both open and closed hihat sounds), two identical and very versatile synth percussion channels and finally a cymbal channel. Below, you will find a detailed description of each of these Instruments.

The equalizer (**EQ**) is an integral to the Heartbeat sound and should be thought of as part of the drum sound.

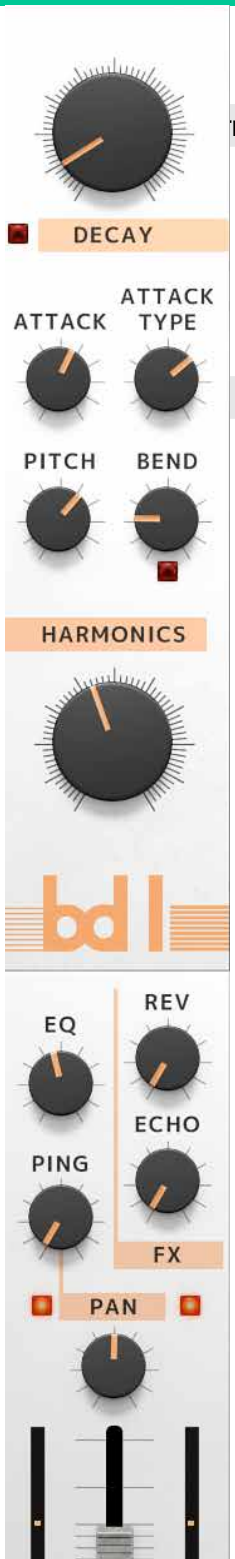
1. Bass drums
2. Snare drums
3. Hihat
4. Percussion
5. Cymbal

Bass Drum 1 "BD 1"

BASS DRUM 1 is highly flexible and was inspired by a well known Japanese drum machine from 1984 that went more or less unnoticed until the end of the 80s, when it became the core of the new house music scene of Chicago, Detroit and New York. Its sound stems from a modeled analog synthesized tone with a slight drop in pitch in its decay, augmented with a waveform attack transient.

Decay	Sets the duration of the bass drum sound. Turn counter-clockwise for short popping sounds and clockwise for longer ones.
Attack	Sets the level of the attack transient waveform.
Attack Type	Sets the character of the attack transient. Turn counter-clockwise for electronic style, harsher sounds, and clockwise for more "woody" and acoustic sounds.
Pitch	Sets the initial pitch of the modeled analog synthesis.
Bend	Turn clockwise for a fast pitch bend that goes up and then down again. Set fully counter-clockwise to bypass.

To create short percussion-like sounds , set the **Eq** counter-clockwise to remove the bottom end from the bass drum, .



monics	Adds harmonics/distortion to the synthesized tone. Can go from subtle overtones into harsh and bit-crunchy territory above the 12 o'clock position.
EQ	Boosts or cuts the low frequencies of the bass drum.

House music kick drum

Characteristic of the house music kick drum is its short and distinct snappy attack along with its moderate decay. By changing **Decay**, **Bend** and **Harmonic** you'll get different and useful variations.

Decay: 25%
Attack: 50%
Attack Type: 0%
Pitch: pretty much what ever you like, but 0% will do.
Bend: 50%
Harmonics: anywhere between 0% to 20% will do.
EQ: 30%

Acoustic style kick drum

The acoustic bass drum is short and dry. Decrease the **Attack** volume if you want the impact to be a bit smoother.

Decay: 10%
Attack: 100%
Attack Type: 71%
Pitch: -75%
Bend: 84%
Harmonics: 0% (clean)
EQ: 0%

Bass Drum 2 "Kick"

The second bass drum is circuit modeled from a classic Japanese drum machine from the early 1980s. It has been heavily used in many genres, ranging from electro and hip-hop to techno and R&B.

Decay Sets the duration of the bass drum sound. Turn counterclockwise for short popping sounds and clockwise for longer ones.

Attack Adjusts the filter level of initial click transient. Turn counterclockwise for a darker and more subdued click character, and clockwise for a more edgy and apparent click.

Pitch Sets the pitch of the bass drum sound.

Harmonics Sets the amount of clipping distortion.

EQ Boosts or cuts the low frequencies of the bass drum.

By using **Velocity To Pitch** in combination with this bass drum it is possible to create nice sounding deep baselines.

Electronic style booming kick drum

Bass Drum 2 is very suited for this type of booming electronic sounds with long Decay times.

Decay: 100%
Attack: 20%
Pitch: -32%
Harmonics: 0% (clean)
EQ: 35%

Techno style kick drum

Short and distinct kick that will cut through any mix.

Decay: 10%
Attack: 100%
Pitch: -45%
Harmonics: 0% (clean)
EQ: 0%

Hollow distorted kick drum

This setting makes the bass drum sound more like a synth bass.

Decay: 73%
Attack: 100%
Pitch: 100%
Harmonics: 100% (clean)
EQ: -100%

Snare/Rimshot "SD RIM"

The SNARE/RIMSHOT channel blends snare drum waveforms with modeled analog synthesis in a highly flexible manner. The balance between them is determined by the slider at the top, and the WAVE and SYNTH portions each have three knobs that adjust their respective sound character—green for WAVE and blue for SYNTH.

Wave/Synth	Sets the balance between the waveform and synthesized portion of the drum sound.
Type	Sets the character of the sound. WAVE (green knob) ranges from a rattling snare to a hard rimshot, SYNTH (blue knob) takes the sound from a pitched note to a noise sound.
Pitch	Sets the pitch for each sound, and the cutoff for the noise in the synthesized part.
Decay	Sets the duration of the WAVE and SYNTH parts respectively. Turn counterclockwise for short popping sounds and clockwise for longer ones.
EQ	Boosts or cuts the frequency range where most of the snare drum's tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the snare/noise character) and clockwise to boost (emphasizing the tonal character).

House snare

A snare sound close to that of a very popular drum machine from the 80s.

Wave/Synth: -42%
Wave type: -34%
Synth type: 0% (TONE)
Wave pitch: 0%
Synth pitch: -27%
Wave decay: 100%
Synth decay: 21%
EQ: 39%

Acoustic Style Snare

A more acoustic sounding snare drum.

Wave/Synth: -13%
Wave type: -60%
Synth type: -57%
Wave pitch: -52%
Synth pitch: -55%
Wave decay: 77%
Synth decay: 72%
EQ: -21%

Create dynamic and interesting sounds by combining a wave portion with short decay with a synth portion with long decay, and vice versa.

Snare/Clap "SD CLP"

The second snare drum is the SNARE/CLAP and works similarly to the SNARE/RIMSHOT. However, both the WAVE and SYNTH portions of the SNARE/CLAP have a different sound character which lends itself more to clap style sounds.

Create a double-clap sound by using a clap wave combined with a synthesized clap.

House Clap

A clap sound close to that of a very popular drum machine from the 80s.

Wave/Synth: -80%
Wave type: 100% (CLAP)
Synth type: 100% (CLAP)
Wave pitch: -7%
Synth pitch: 80%
Wave decay: 100%
Synth decay: 57%
EQ: 71%

Drummachine Snare

An 80s style digital sounding snare drum.

Wave/Synth: -64%
Wave type: -35%
Synth type: 0% (SNARE)
Wave pitch: 0%
Synth pitch: -20%
Wave decay: 64%
Synth decay: 36%
EQ: 76%

Wave/Synth	Sets the balance between the waveform and synthesized portion of the drum sound.
Type	Sets the character of the sound. WAVE (green knob) ranges from different snare sounds to tight claps, while SYNTH (blue knob) takes the sound from a slightly noisy tonal character to a dark and sluggish clap sound.
Pitch	Sets the pitch for each sound, and the cutoff for the noise in the synthesized part.
Decay	Sets the duration of the WAVE and SYNTH parts respectively. Turn counterclockwise for short popping sounds and clockwise for longer ones. For the SYNTH portion of the sound, longer **Decay** times also decreases the tightness of the clap, which gives it an even more loose and sluggish character.
EQ	Boosts or cuts the frequency range where most of the snare drum's tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the snare/noise character) and clockwise to boost (emphasizing the tonal character).

Hihats

The HIHATS instrument of Heartbeat can make both closed and open hi-hat sounds. One sound chokes the other. So if the open hi-hat is played, followed by the closed hi-hat the open hi-hat will be immediately cut off by the closed hi-hat. The HI-HATS are laid out in the same way as the two snare drums (SNARE/RIMSHOT and SNARE/CLAP), with a crossfade slider at the top which sets the balance between hi-hat waveforms (WAVE), and sounds generated by modeled analog synthesis (SYNTH). As with the snare drums, the green knobs affect the WAVE portion and the blue knobs affect the SYNTH portion of the sound.

Wave/Synth	Sets the balance between the waveform and synthesized portion of the hi-hat sound.
Type	Sets the character of the sound. WAVE (green knob) ranges from classic drum machine-like hi-hat sounds to a more acoustic sounding character. SYNTH (blue knob) takes the sound from a cluster of high pass filtered pulse waveforms in the far counter-clockwise setting, to a filtered white noise when turned clockwise.
Pitch	Sets the pitch for the WAVE sound (green knob), and sweeps a low cut filter for the SYNTH portion (blue knob).
Decay	Sets the duration of the WAVE and SYNTH parts respectively. Turn counterclockwise for short popping sounds and clockwise for longer ones.
EQ	Boosts or cuts the frequency range where most of the hi-hat's tonal content is found. The neutral setting is at 12 o'clock, turn counterclock-wise to cut this frequency range (emphasizing the noise character) and clockwise to boost (emphasizing the tonal character).

Combine the transient of the wave with a longer, noisy, decay of the synth part to get dirty nice hi-hat sounds.

Synthesized Hi-hat

A sound similar to that of a very popular early 80s drum machine.

Wave/Synth: 93%
Wave type: -45%
Synth type: 52%
Wave pitch: 57%
Synth pitch: 68%
Wave decay: 6%
Synth decay: 24%
EQ: 86%

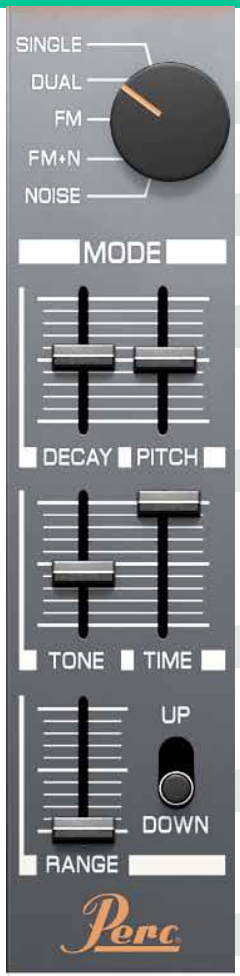
Digital Hi-hat

Dry 80s drum machine style hihat.

Wave/Synth: 33%
Wave type: 3%
Synth type: 100% (NOISE)
Wave pitch: -4%
Synth pitch: 74%
Wave decay: 66%
Synth decay: 6%
EQ: -49%

Percussion 1 and 2 "Perc"

The two PERCUSSION instruments are identical to each other, so the information given here covers them both. They are highly flexible instruments that draws a lot of inspiration from the lesser known, but very powerful, Japanese synth percussion unit from the early 80s. The sound of the PERCUSSION 1 and 2 are purely generated from modeled analog synthesis.



Mode — This dial selects one of the following five modes.

SINGLE: Employs a single triangle wave oscillator. This is great for disco style toms, additional bass drums and short harmonic snaps.

DUAL: Employs two triangle wave oscillators with a fixed pitch ratio between them. This is ideal for cowbell, agogo bell and marimba type of sounds.

FM: One oscillator is frequency modulated by the other with a fixed ratio. This is useful for disharmonic metal-like sounds.

FM+N: The same as above, but with added noise. Can be used to generate otherworldly metallic sounds.

NOISE: White noise. This is good for generating shakers, thunderous snares and special effects.

Sets the duration of the sound.

Sets the initial pitch of the oscillators.

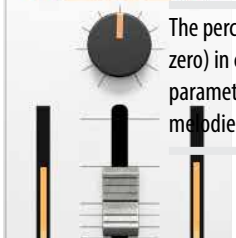
Sets the initial cutoff frequency of the low-pass filter.

Sets the speed of the pitch bend and in noise mode the speed of the filter-sweep.

Sets the amount of pitch bend or filter-sweep in noise-mode.

Sets if the bend goes upwards or downwards (filter sweep in noise mode).

Boosts or cuts the frequency range where most of the drums tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the noise character) and clockwise to boost (emphasizing the tonal character).



The percussion modules without the bend (**Range** at zero) in combination with the **Velocity To Pitch** parameter makes it possible to create small pseudo-melodies and baselines.

Cymbal

The sound of Heartbeat's Cymbal is purely generated by modeled analog synthesis. It draws inspiration from several early 80s Japanese drum machines. But the **Ring** parameter has been added for the ability to get a more bell-like high pitched ringing sound.

Decay Sets the duration of the sound.

Ring Sets the amount of "ring" character.

Pitch Adjusts the filtered mix of the harmonics within the cymbal sound.

EQ Boosts or cuts the frequency range where most of the cymbals tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the noise character) and clockwise to boost (emphasizing the tonal character).

A high setting of **Ring** makes the cymbal sound more like a ride cymbal, while a low setting makes the cymbal more vintage drum box sounding.

Noise Hat

Using the cymbal as an extra noisy drum-machine like hihat.

Decay: 0%
Ring: 0%
Pitch: 100%
EQ: 100%

Noise Ride

A cymbal sound close to that of a very popular drum machine from the 80s.

Decay: 14%
Ring: 10%
Pitch: 40%
EQ: 48%

Short and Sweet

Short cymbal sound with some ringing.

Decay: 20%
Ring: 34%
Pitch: 53%
EQ: -47%

The Mixer

The Mixer section takes up most of the lower left part of the Heartbeat graphical user interface. The parameters are identical for all eight mixer channels, with the exception of the **EQ** (equalizer) which is tuned for each individual instruments, although the knobs look identical.

EQ Adjusts the equalizer setting. It is tuned for each channel and optimized to work with the sweet spots of the individual instruments.

Rev Sets the signal level being sent from each instrument to the TSAR-1D REVERB unit, and therefore how much reverb is added to the instrument. The **Rev** send is post-fader, meaning that the send level is also affected by the setting of the **Volume** fader. This keeps the proportion between the direct sound and the reverb intact even if the **Volume** fader is turned up or down.

Echo Sets the signal level being sent from each instrument to the FILTER ECHO, and therefore how much delay is added to the instrument. The **Echo** send is post-fader, meaning that the send level is also affected by the setting of the **Volume** fader. This keeps the proportion between the direct sound and the reverb intact even if the **Volume** fader is turned up or down.

Ping/Pong The automated panning function. Sets the amount of automatic panning for each drum hit.

Pan The initial position of the instrument in the stereo panorama.

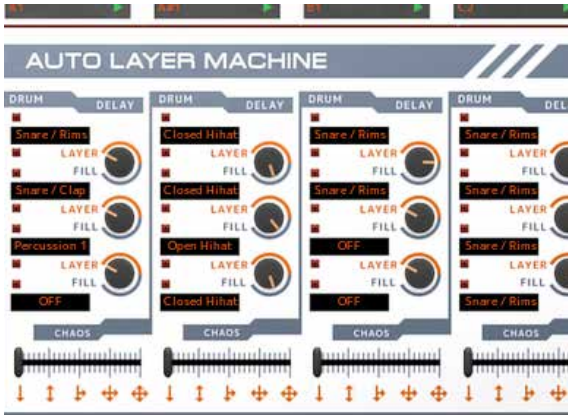
Volume fader Sets the volume of the instrument.

Solo (S) Activating SOLO for a mixer channel mutes all othter channels (unless they are also in SOLO mode).

Mute (M) Activating MUTE for a mixer channel turns off the sound from this channel.

Auto Layer Machine

AUTO LAYER MACHINE takes up most of the upper right part of the graphical user interface. It can be used to easily layer sounds from two or more instruments for new textures, or to trigger flams or auto-fills—you could compare it to a basic MIDI sequencer. AUTO LAYER MACHINE consists of four channels, each with its individual MIDI note number assigned. The four channels are divided into a **Drum** and a **Delay** column. In the **Drum** column you will find four slots per channel. Hitting the assigned MIDI key will make the corresponding AUTO LAYER MACHINE channel generate a chain of events, moving from the top slot to the bottom one (in its default state). Each trigger will be slightly lower in velocity which is apparent when using the **Velocity** parameters.



Get started!

The easiest way to understand what the AUTO LAYER MACHINE does is by trying the settings in the factory preset you will have every time you launch a new instance of Heartbeat. Here, the four AUTO LAYER MACHINE channels are set up to perform different tasks.

Layering

Hit the MIDI key A1 to trigger the first AUTO LAYER MACHINE channel (or click its green arrow in the UTILITY section, the black field above the channel). You will hear that this triggers three of the instruments—SNARE/RIMSHOT, SNARE/CLAP and PERCUSSION 1—simultaneously. This creates a layered sound. You can also see the names of these three instruments in three of the slot windows of the first AUTO LAYER MACHINE channel, indicating that the slots have been assigned to these instruments.

Patterns and fills

If you instead trigger the second AUTO LAYER MACHINE channel, by hitting A#1, you will hear the closed and open HIHATS playing a short pattern with four hits. Again, you can see in the slots that they are assigned to the closed and open HIHATS. But unlike the first channel, they were not playing simultaneously—a delay was added for each step.

This is done with the knobs in the **Delay** column, to the right of each slot. The **Delay** knobs set how long it takes after a slot has been triggered until it passes on the trigger impulse to the slot below it. In the first AUTO LAYER MACHINE channel, you will see that the **Delay** knobs are all set to LAYER (fully counter clockwise), meaning that there is no delay from one slot to the next—the instruments are triggered simultaneously.

But in the second channel, they have other positions, which is what creates the delay between the instruments being triggered, and thus creates the small pattern you hear each time you trigger the channel. If you change the positions of the **Delay** knobs, you will hear the short pattern change accordingly.

Velocity

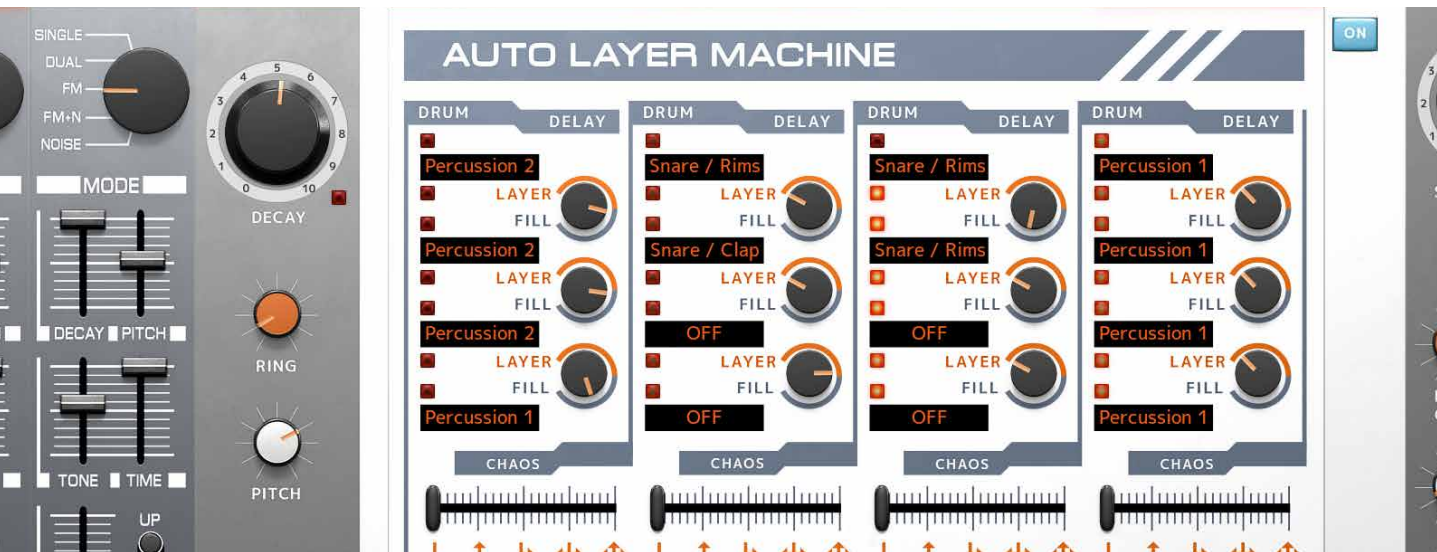
The Instrument in the first Auto Layer Machine slot will be triggered with the velocity of the incoming MIDI note. For each subsequent step in the Auto Layer Machine, the velocity will automatically drop by a predefined amount. This means that if the incoming MIDI note has a very low velocity to begin with, the subsequent steps might drop below 0 velocity, and thus not trigger the Instrument at all.

On Turns ON and OFF the Auto Layer Machine. You can save some CPU power by turning the Auto Layer Machine off when not in use.

Slot window The slot windows in the **Drum** column indicates and determines which of Heartbeat's instruments is triggered via the slot. Click or SHIFT-click to select next or previous instrument. You can bypass the slot entirely by selecting OFF. It is also possible to click and drag to scroll back and forth among the instruments.

Delay Determines how long it takes after an instrument has been triggered until it passes on the trigger impulse to the slot below it. By setting it to ZERO (the knob indicator pointing at LAYER), there is no delay, so the two instruments are triggered at the same time and thus layered. By turning it clockwise, the following trigger will be more delayed. Use this to create flams or automatic fill patterns. When the indicator is by the orange part of the marking, the delay is expressed in milliseconds in the tooltip window that pops up. Turn it to the blue side to set the delay expressed as beat divisions of the DAW project's tempo.

Chaos In its default state with the slider set all the way to the left, the trigger impulses move from the top to the bottom as indicated by the orange arrow underneath the Chaos slider. Moving the slider a bit to the right will enable the Auto Layer Machine to reverse the direction of the triggers so that some triggers will randomly populate upwards instead of only downwards. Moving the slider even further will make the trigger impulses "spill" over to the adjacent Auto Layer Machine channels, as indicated by the arrows. In its far right position, you will have full chaos with triggers sent everywhere in a rather unpredictable manner. Even more so if you have all four Chaos sliders to the far right!



The Effects

Heartbeat includes three different effect units: The VALLEY PEOPLE DYNA-MITE compressor/limiter/gate, the TSAR-1D REVERB and the FILTER ECHO. These are shared by all the drums. The signal level sent from each drum to the TSAR-1D and the FILTER ECHO is set with the **Rev** and **Echo** knobs in each mixer section. VALLEY PEOPLE DYNA-MITE is inserted across the stereo sum of all the instruments, so as long as it's activated it affects all instruments (apart from the bass drums when **Duck** is set to BASS DRUMS).

Valley People Dyna-mite

The VALLEY PEOPLE DYNA-MITE built into Heartbeat is a specially adapted version of the Valley People Dyna-mite plug-in, separately available from Softube. The original analog Valley People Dyna-mite unit came out in the early 80s and was a very popular tool for gating/expanding, compressing/limiting and ducking—highly loved for its ability to compress sounds with fierce aggression, and to gate in an ultra-musical manner. The Heartbeat version features four operating modes:

Compression

A compressor is basically an automatic volume control, which turns down strong sounds but leaves the weaker sounds unaffected. This makes the dynamic range of the sound (the difference between strong and weak parts) smaller, which is why it's called compression. Using compression lightly can make the sound compact and coherent (often described as glueing the sounds together), while using it heavily can create an aggressive mash. With the **Mode** switch set to SLOW, the Dyna-mite will act as a compressor with a slow attack. This lets the initial transient of the sound through before Dyna-mite reacts and starts compressing, resulting in a punchy and snappy sound. The **Threshold** knob sets the threshold level. Any time a sound reaches above this level, Dyna-mite will start compressing—so a high threshold setting will only affect the strongest peaks, while a low threshold setting will affect most of the incoming sound, resulting in a very apparent compression. The **Release** knob determines the time it takes for the Dyna-mite to recover after it has compressed. This can be used to emphasize the rhythmic feel of the beat, making the Dyna-mite "breathe" in time with the music.

Limiting

A limiter is a very fast compressor that uncompromisingly slams down the sound any time it exceeds the threshold level. Its original use was to protect loudspeakers from sharp sound spikes that could potentially damage them, but it can also be used creatively for music mixing. Set Dyna-mite's **Mode** switch to FAST, and it will act like a limiter. Compared to the SLOW mode, you will note that the Dyna-mite now doesn't let the initial transient of the sound through. Instead, the sound hits a brick wall, creating an aggressive and pumping sound—even more so with a low threshold setting.

Gating

With the **Mode** switch set to GATE, Dyna-mite will shut off the sound completely if it drops below the threshold level (set by the **Threshold** knob), and open up as soon as the sound rises above the same level. This can be used to make sounds appear shorter (for example

creating gated reverb effects), and to get rid of low-level sounds for a cleaner and more focused impression. If the Dyna-mite is set to gate out the weaker sounds entirely, the **Range** knob can be used to mix them back in, but at a lower level than they originally had. This is called *expansion*—you expand the dynamic range of the mix by making the weak sounds (the ones below the threshold level) weaker, and thus in comparison making the strong sounds (above the threshold level) stronger. So an expander is basically a compressor in reverse. This can be used to enhance and alter the dynamic feel of a drum beat.

Ducking

Ducking is the effect of one sound source controlling the output volume of another. In its Heartbeat version, Dyna-mite can be set to let the bass drum channels duck the others. By setting the **Duck** switch to BASS DRUMS and the **Mode** switch to either FAST or SLOW, every time a hit from one of Heartbeat's bass drums is strong enough to reach above the threshold level, the sound level of all the other instruments will be turned down by Dyna-mite. This creates a pumping and energetic effect that is prominently used in a lot of electronic dance music. The **Mode** switch can also be set to GATE while the **Duck** switch is set to BASS DRUMS. In this case, the gate opens up every time the bass drum hits, and shuts off the sound of the other instruments between the bass drum hits.

On Turns Dyna-mite on or off.

Presets Clicking the button with the white arrow below the On button brings up Dyna-mite's preset menu. It contains some examples of applications of Dyna-mite compressor which are good starting points for further tweaking.

Threshold Sets the threshold level, above which the Dyna-mite starts to limit or compress (in FAST or SLOW modes), or lets the sound through (in GATE mode).

Release Adjusts the time it takes to restore the original gain after gating/compressing.

Duck Activates/deactivates bass drum ducking, which makes the bass drums affect Dyna-mite's processing of all the other instruments.

Mode Sets the main mode of operation: GATE, FAST (limit) or SLOW (compress).

Range Sets the maximum amount of gain reduction.

Output Sets the output volume. If necessary, turn this up to compensate for the volume loss caused by compressing/limiting.

Filter Echo

Filter Echo is a gritty little delay effect with a resonant lowpass filter in its feedback loop. The filter can be set to near self-oscillation for that lo-fi sound.

Turns off the sound of the Filter Echo.

Solo (S) Solos the sound of the Filter Echo.

Pre Places the Filter Echo before Dyna-mite in the effects chain, which means Filter Echo's sound will also be affected by Dyna-mite's processing. When the Pre button is not activated, the Filter Echo's output will be post the Master Bus Saturation effect, but before the Master Bus equalizer.

Presets Clicking the button with the white arrow below the Pre button brings up Filter Echo's preset menu. It contains some examples of Filter Echo settings that are good starting points for your tweaking.

Time Determines the delay time, how long time passes between each delay "hit". In the left half of the knob's path, the range is from 1 to 1000 milliseconds. In the right side, the delay time can be set in divisions of the DAW tempo, ranging from 1/64 to 1/2 beat. The latter is useful for setting the delay to act in time with the song.

Feedback This set the amount of feedback, how many delay repeats there will be. It ranges from one repetition to roughly 10 repetitions at full feedback.

Cutoff Sets the cutoff frequency of the low-pass filter.

Resonance Sets the resonance of the low-pass filter.

Volume Sets the output volume.

TSAR-1D Reverb

The TSAR-1D is a version of Softube's acclaimed TSAR-1 Reverb, adapted for use with Heartbeat. For more information, please see page 515. TSAR-1 Reverb is available as a separate plug-in from Softube.

Mute (M) Turns off the TSAR-1D.

Solo (S) Solos the TSAR-1D reverb.

Pre Places the TSAR-1D before Dyna-mite in the effect chain, which means that the reverb will also be affected by Dyna-mite. When the **Pre** button is not activated, TSAR-1D's output will be after the master saturation, but before the master equalizer.

Presets Clicking the button with the white arrow below the Pre button brings up TSAR-1D's preset menu.

Predelay Determines the time between the dry signal and the reverb tail. Set to 0, there is no delay. Delayed settings are often used to achieve the impression of a large room, by making the reverb tail arrive later.

Time Sets the duration of the reverb sound, from short to long.

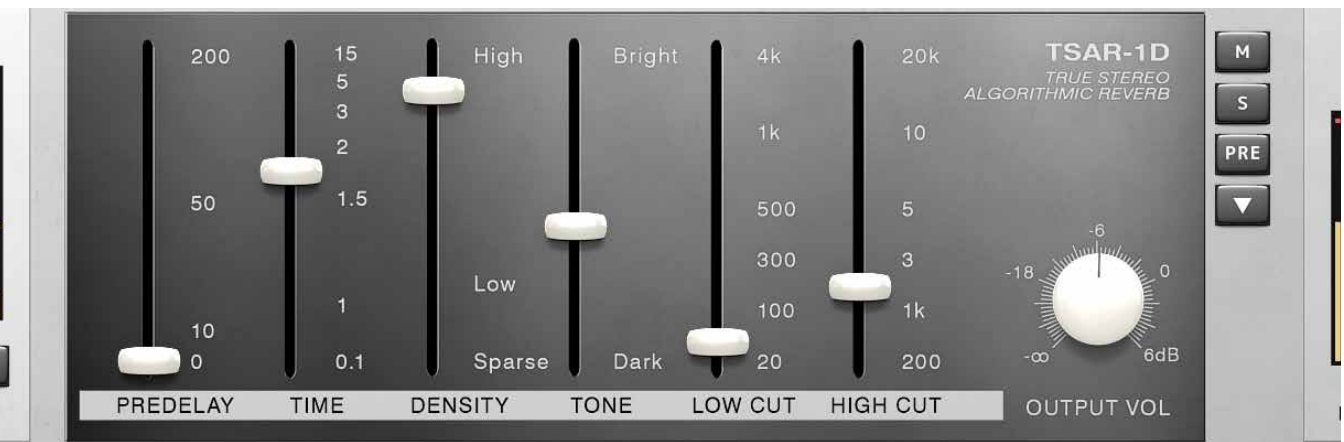
Density Adjusts is the thickness and smoothness of the reverb.

Tone Overall tone of the reverb signal.

Low Cut Applies a low cut filter to the reverb sound, taking away the lower frequencies that might make the sound cluttered and undefined.

High Cut Applies a high cut filter to the reverb sound, taking away the higher frequencies that might make the reverb sound take up too much space in the mix.

Output Vol Output volume of the reverb.



Master Bus

The Master Bus is the section on the far right of Heartbeat's graphical user interface. This affects the main output of Heartbeat, letting you add saturation, make EQ adjustments to the overall sound and more.

Saturation	Sets the amount of saturation applied to the entire drum mix, post the Dyna-mite compressor. It mimics the saturation that can be achieved by devices using electronic vacuum tubes, which results in a pleasing and thick saturation.
High Cut	Applies a high cut filter to Heartbeat's output, which cuts treble frequencies and makes the overall sound darker. This is very similar to the High Cut function of a DJ mixer.
High Gain	Boosts the treble frequencies.
Mid Freq	Sets the center frequency of the midrange equalizer filter.
Mid Gain	Boosts the midrange at the frequency determined by the Mid Freq knob.
Low Gain	Boosts low frequencies.
Low Cut	Applies a low cut filter, which cuts bass frequencies and makes the overall sound thinner. This is very similar to the Low Cut function of a DJ mixer.
Mono Cut	Determines a cutoff frequency below which everything is summed to mono. This is a great way to ensure that your mix will sound solid on any playback system, since it guarantees that you will have no phase issues in the low end of the frequency spectrum.
Width	At 12 o'clock, Width is disabled and all stereo settings work as expected. Turning it counterclockwise gradually makes the sound become more mono. Turning it clockwise makes the sound wider.
aster Out	Master fader which controls the volume of Heartbeat's main output.

Velocity

At the lower left of Heartbeat's graphical user interface, below the mixer section, are the global **Velocity** parameters. These knobs determine how responsive Heartbeat will be to the velocity of the incoming MIDI signals—i.e., how strong or soft the incoming MIDI note is. The **Velocity** settings are global for Heartbeat's instruments, meaning that they effect them all simultaneously.

Velocity Sets how strongly the velocity of the incoming MIDI note affects the volume of the instrument being triggered. Setting this to 0% (fully counterclockwise) will result in no volume difference of the sounds, regardless of the velocity. Conversely, when **Velocity** is set to 100% (fully clockwise), the Instruments will respond very dynamically to the velocity. So in this setting, higher velocities give louder sounds.

To Pitch Sets how the initial pitch of Heartbeat's instruments is affected by velocity. In its 12 o'clock position, velocity has no effect on pitch. Turning it counterclockwise will result in higher velocities giving the sounds a lower initial pitch. Conversely, turning it clockwise will result in higher velocities giving the sounds a higher initial pitch.

To Attack Sets the amplitude of the attack portion of the instrument's sounds. In its 12 o'clock position, velocity has no effect on attack levels. Turning it counterclockwise will result in higher velocities giving lower attack amplitude. Conversely, turning it clockwise will result in higher velocities giving the sounds a higher attack amplitude. This applies only to instruments that have the **Attack** parameter

To Decay Sets how the decay time of Heartbeat's instruments is affected by velocity. In its 12 o'clock position, velocity has no effect on decay time. Turning it counterclockwise will result in higher velocities giving the sounds shorter decay times. Conversely, turning it clockwise will result in higher velocities giving the sounds longer decay times.

Time Gate

Time Gate is a fun and useful function that cuts the decay short of all instruments globally. This creates a jerky and chopped up cool sound reminiscent of old 80s sample-based drum machines with very small memory. Since Time Gate is controlled by a single knob, it offers a quick way of altering the sound of the entire beat—this is not least useful for live applications.

Using Multiple Outputs

Heartbeat is designed to be a one-stop shop for drum sound creation, where the resulting sound package comes out of a single stereo output. But for added flexibility, it is also possible to send the individual instruments through separate outputs, and have them appear on individual mixer channels in your DAW. The separate outputs can be used in situations where you would like to add mix effects from your DAW to the individual instrument sounds of Heartbeat—for example if you have a particular reverb plug-in in your DAW that you would like to use for the snare drums, and only for the snare drums. Or if you want Heartbeat's bass drums to duck all the other sound sources in your song, such as synths and vocals.

When using separate outputs, the signal from the instrument is being split into two. One is sent the usual way through Heartbeat, via the volume fader and the effects to Heartbeat's Master Bus. The other one is sent to the direct output. This is tapped out of the DRUM CHANNEL mixer pre-fader. This means that the DRUM CHANNEL volume fader (as well as the **Solo** and **Mute** buttons) will *not* affect the signal being fed to the direct output. If you want an instrument sound to only be sent to the direct output, and not appear in your main Heartbeat mix at the same time, you can set the corresponding DRUM CHANNEL's volume fader to zero, or press its **Mute** button.

Sending effects to a separate output

If you want all the instruments on separate outputs, and also get the send effects (FILTER ECHO and TSAR-1D) as a separate stereo signal, in total 8 + 1 stereo pairs, you can achieve this using the **Solo** buttons for the FILTER ECHO and TSAR-1D—then you will only have the outputs of these sent to Heartbeat's MASTER BUS.

Using multiple outputs

Your DAW will automatically detect that Heartbeat has a total of nine stereo outputs—the MAIN OUTPUT, plus a stereo pair for each of the eight instruments. These outputs are named in accordance with the DRUM CHANNELS:

- BASS DRUM 1
- BASS DRUM 2
- SNARE RIMSHOT
- SNARE CLAP
- HIHATS
- PERCUSSION 1
- PERCUSSION 2
- CYMBAL

The different DAWs all have their own particular ways of handling instruments with multiple outputs, such as Heartbeat. Therefore, please refer to your DAW's manual to learn how to use Heartbeat's multiple outputs on your particular system.

Presets

Heartbeat features 50 different preset drum kits, ranging from classic drum machine sound-a-likes to more contemporary sounds of all kinds. They also contain settings for the effects as well as programmed Auto Layer Machine settings. Each preset name begins with a two-letter acronym of the name of the creator. They are:

KU	Kristofer Ulfves, Softube
CB	Christoffer Berg (Depeche Mode, The Knife, Hird)
DG	David Giese (Joxaren, Flogsta Danshall)
TB	Tomas Boden (Differnet, Liminals)

Setup window

Clicking the **Setup** tab in the bottom right corner of Heartbeat's graphical user interface will bring up the setup configuration menu. Some of the changes made here will only take effect after relaunching Heartbeat.

Always Use Small GUI	This toggles between bigger and smaller versions of the graphical user interface. We recommend that you check this box if you use Heartbeat on a small computer screen, such as a laptop screen.
Enable Tooltip	Toggles Tooltips on and off. These are the small pop-up windows that appear when hovering the mouse pointer over most of Heartbeat's controls.
Show Value Display	Toggles the value display in lower left corner of Heartbeat on and off.

Credits

Oscar Öberg – product lead and signal processing. **Kristofer Ulfves** – research, sound design, presets and user manual. **Niklas Odelholm** – graphic design and presets. **Patrik Holmström** – GUI programming. **Henrik Andersson Vogel** – user manual and marketing. **Paul Shyrinskykh** – quality assurance. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – graphics rendering. **Torsten Gatu** – framework programming. **Mattias Danielsson** – technical support. **Johan F. Antoni** – help with initial concept. **Andreas Tilliander** – hardware reference. **Tomas Boden** – testing and presets. **Christoffer Berg** – testing and presets. **David Giese** – testing and presets. **Jakob Herrman** – sound reference. **Marcus Schmahl** – demo and feedback

Block Diagram

This is a simplified block diagram of the Heartbeat functionality and signal paths.

MIDI notes are received by the Auto Layer Machine and the Drum Channels. Audio from the Drum Channels are routed both to the separate outputs as well as to the Valley People Dyna-mite.

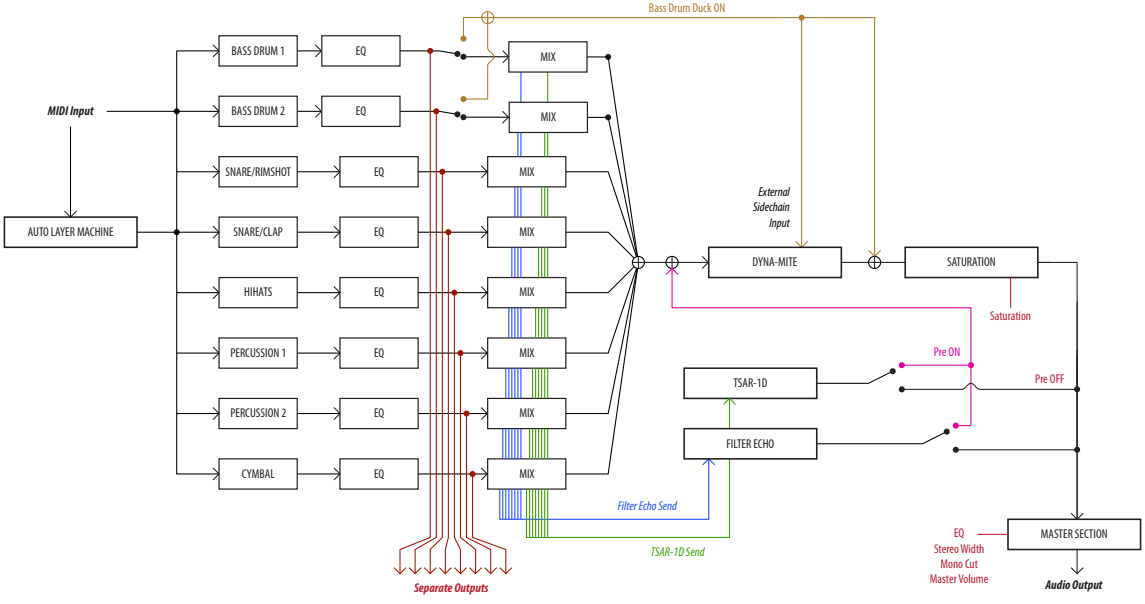
Bass Drum Ducking

If Dyna-mite's **Duck** switch is set to BASS DRUMS, the audio from the bass drums are being routed to the external sidechain of the Dyna-mite, as well as being mixed together with the output of the Dyna-mite.

Send Effect Pre/Post

If the send effects (TSAR-1 Reverb and Filter Echo) are set to PRE (**Pre** button is blue), the outputs from the effects are routed to the input of the compressor.

If the send effects' **Pre** button is off (**Pre** button is gray), the output from the effects are routed directly to the master section, but after the saturation circuit.



ALM/Filter Echo times chart

Name	Length	Swing value	
1/64	64th note		
1/32	32th note		
1/16	16th note		
1/16+	16th note	54% swing	
1/8T-	slightly short 8th note triplet	16th note with 62% swing	
1/8T	8th triplet	16th note with 66% swing	
1/16D	dotted 16th note	16th note with 75% swing	length of a 16th plus a 32th note
1/8	8th note		
1/8+	8th note	54% swing	
1/4T-	slightly short quarter note triplet	8th note with 62% swing	
1/4T	quarter note triplet	8th note with 66% swing	
1/8D	dotted 8th note	8th note with 75% swing	length of a 8th plus a 16th note
1/4	quarter note		
1/2T	half note triplet	4th note with 66% swing	
1/4D	dotted quarter note	4th note with 75% swing	length of a quarter plus a 8th note
1/2	half note		

MODULAR EXCLUSIVE

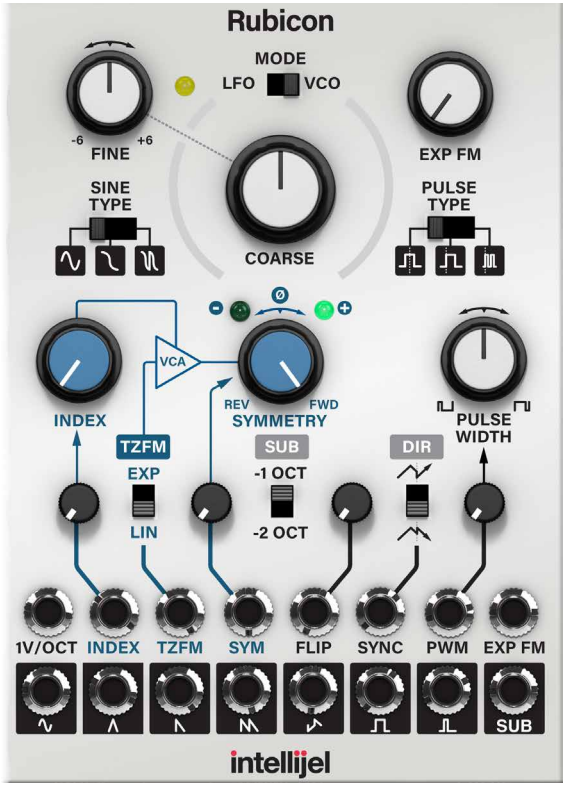
29 Intellijel Rubicon, Korgasmatron II & uFold II

Eurorack brand Intellijel, run by **Danjel van Tijn** and associates, is based in Vancouver, British Columbia in Canada. These branded add-on modules from Intellijel are for purchase at `http://softube.com/buy`.

The Intellijel modules are RUBICON THROUGH ZERO OSCILLATOR, KORGASMATRON II and the µFOLD II. These three modules all make a fine addition to the more traditional subtractive basic system of Softube Modular, as well as a step towards more west-coast thinking in terms of synthesis.

Intellijel Rubicon

The Rubicon is David Dixon's masterpiece bringing a new approach on thru-zero frequency modulation into the Eurorack world. Rubicon exclusive tone and extreme tracking (0.1% over 8 octaves) is ideal for experimenting with FM synthesis.



Parameters

Fine This is the fine tuning knob, ranging up and down six semitones.

Exp FM This knob sets the amount of exponential frequency modulation to be applied by the signal coming from **Exp FM** input jack (see below).

Sine Type This switch changes the shape of the sine output jack between sine, sigmoid and double sigmoid.

Coarse This knob is coarse tune that ranges from a few Hz to above hearing range.

Pulse Type This switch determines the shape and phase of the **Pulse** output jack. It switches between CENTERED, EDGE ALIGNED and EDGE ALIGNED, DOUBLE FREQUENCY.

Index This is the offset gain of the signal connected to the through zero **FM (TZFM)** jack.

Symmetry This sets the symmetry offset of the output waveforms. Normal forward position is preset to the right, center position is 0% which means waveforms are slowed down to a stop. Left position is completely reversed. Note that **Symmetry** affects 1V/Oct tracking! For the best tracking keep the **Symmetry** knob in full clockwise position.

Pulse This knob set the pulse width offset for the pulse output.

[index CV amount] This knob sets the index CV amount of the incoming signal at the index jack affecting the amplitude of the through zero frequency modulating signal.

TZFM Exp/Lin Switch which determines the response of the through zero frequency modulation, exponential or linear.

[symmetry CV amount] This knob sets the symmetry CV amount of the incoming signal at the symmetry jack.

SUB This switch determines whether the sub output will be one or two octaves below Rubicons main pitch.

Soft Sync CV This attenuator is to adjust the incoming signal at the soft sync input jack (flip).

Dir Reset direction. This switch selects whether the hard sync input will reset the VCO to a rising or falling direction.

PWM CV This knob controls the amount of pulse width modulation on pulse waveform by incoming CV at PWM jack.

Inputs

1V/oct CV control of pitch, full range 1V/octave. Connect your note CV from MIDI TO CV converter module to this jack in order to get your Rubicon to track pitch.

Index The incoming signal at the index jack affects the amplitude of the through zero frequency modulating signal.

TZFM The signal input here, through zero frequency modulates the Rubicons waveforms. Connected with **TZFM Exp/Lin** switch described above.

Sym The CV signal input modulates the symmetry setting, and is equivalent to turning the symmetry knob back and forth.

Flip This is the soft sync input. The triangle core of the Rubicon reverses on the incoming sync pulse and the attenuator sets the probability that a pulse will cause reversing. This jack requires edge-y waveforms like square or sawtooth (sawtooth is best). There is a mini attenuator on this input (see above).

Sync Hard sync jack input. Use the reset switch (Dir, described above) to select if the waves will reset on rising or falling edge.

PWM CV input controlling the pulse width of the pulse output.

Exp FM Exponential CV input of the Rubicon. Fully clockwise is 1v/octave tracking.

Outputs

[sine] Sinewave output, but also of the sigmoid and double sigmoid if the Sine type switch is used.

[triangle] Triangle wave output.

[saw wave] Saw wave output.

[double saw wave] Saw wave output one octave above Rubicon fundamental frequency.

[zig-zag] Output of the zigzag wave, a cross between upwards and downward sawtooth.

[square] Square wave output. This output is unaffected by PWM.

[pulse] Pulse wave output.

[sub] Sub oscillator output. This output is unaffected by PWM.

In Use

The Intellijel Rubicon can flip all waveforms backwards using the symmetry function, all except the zig-zag, strangely enough. Really cool waveforms can be achieved by sweeping symmetry while Rubicon is being hard synced by another oscillator. Also very ear-pleasing is using the **TZFM** input by another oscillator in linear mode to create classic "icy" FM timbres.

Intellijel Korgasmatron II

The Korgasmatron II takes its name from the classic Japanese MS series from the early 80s. While the Korgasmatron II is still as gritty as the original hardware, it is also more flexible than its predecessor. The dual filter channels A and B can be set as 2-pole high pass and low pass in series, just like the original, but can also be set to work in parallel and in a whole variety of other modes.

Parameters

Korgasmatron has two identical channels, A and B, with the same basic functionality in both channels. The parameters below controls the patching and blend between channel A and B.

Xfade	This knob sets the cross-fade offset between channel A and channel B.
Xfade CV Amount	This knob controls the CV amount from the **XFade CV** input.
Xfade Dir.	This switch sets the polarity of the **XFade CV Amount** input. When set in the left position, polarity will be positive (cross fade will go from channel A to channel B). When set in the right position, polarity will be negative (the cross fade will go from channel B to channel A).
[filter configuration]	This switch sets the Korgasmatron II's filter configuration of channel A and B. Set to the upper position for a serial filter configuration (one filter after the other), or the lower position for

a parallel configuration. The serial path is disconnected if INPUT B is used in serial mode

The parameters below are identical in both channel A and B.

Clip Type This switch toggles the channel's clipping type between HARD (upper position) and SOFT (lower position).

Cutoff This is the cutoff frequency

Q This knob controls the Q value (feedback) of the filter. Often called emphasis or resonance.

Q-Drive Adjusts headroom in feedback loop, in effect adjust the volume of the resonance when self-oscillating.

FM1 This knob controls the CV amount from the **FM1 CV** input to **Cutoff** frequency.

FM2 This bipolar knob controls the CV amount from the **FM2 CV** input to **Cutoff** frequency. Center position is zero amount, while turning the knob clockwise from center will yield positive modulation and counterclockwise will yield negative modulation.

Mode This switch sets the filter topology and filter type.

The different modes are described on the next page.

In This is the input volume of the filter.

Inputs

FM1 This is the CV input jack for the **Cutoff** frequency. The **FM1 CV** input jack is connected to **FM1 CV Amount** knob (described above).

FM2 CV input jack for the **Cutoff** frequency. The **FM2 CV** input jack is connected to **FM2 CV Amount** knob (described above).

Xfade CV input jack for cross fade sweep

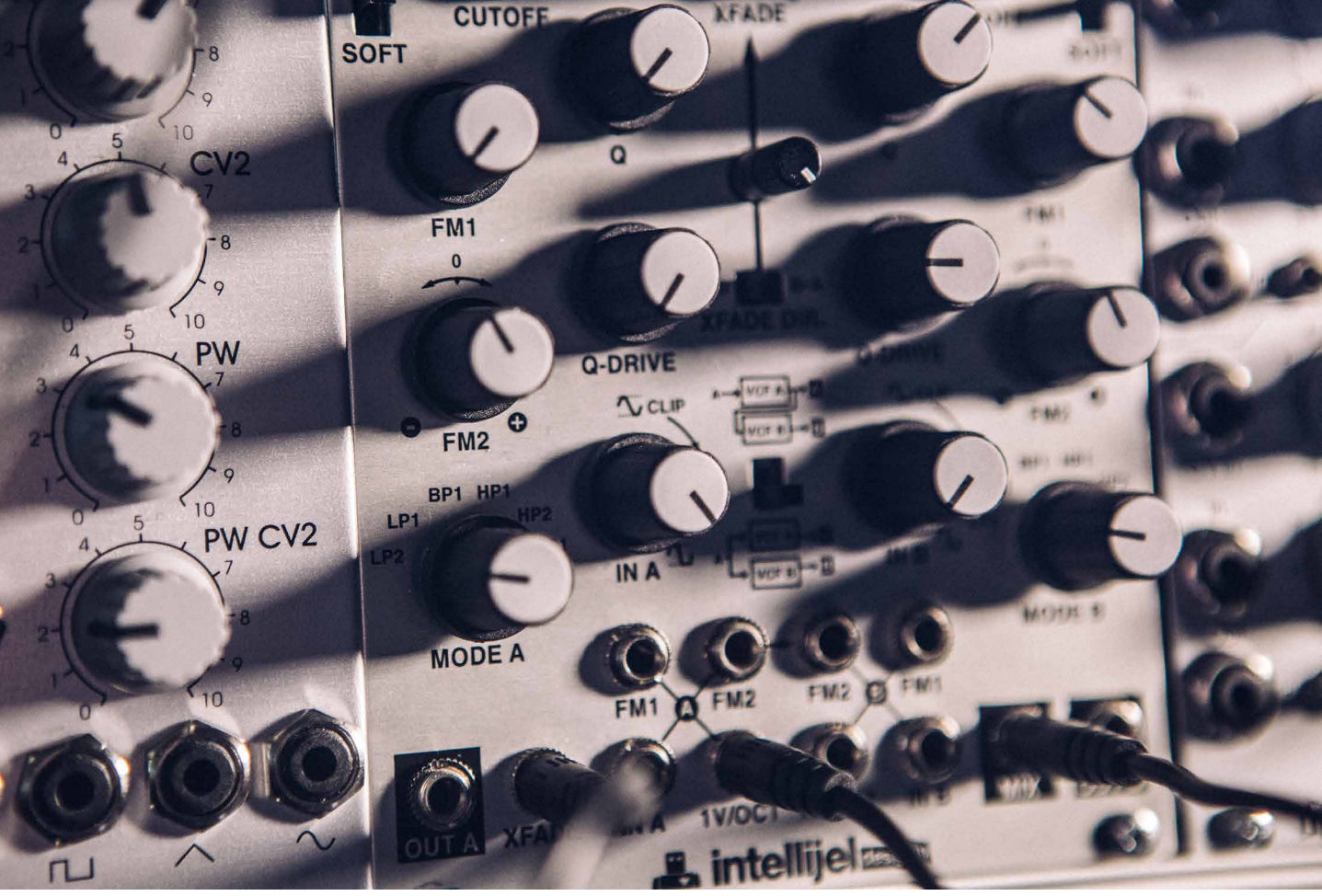
In Audio input jack.

1V/Oct 1V/octave CV-control for the cutoff frequency. Connect your note CV from MIDI TO CV converter module to this jack in order to get your Korgasmatron II to track playable pitch for approximately two octaves (see in use below) .

Outputs

Out Output jack.

Mix Output from both channels, determined by the XFade configuration.



Filter Modes

LP2 2-pole low pass (12dB/octave slope)

LP1 1-pole low pass (6dB/octave slope)

BP1 Band pass, cuts away low and high frequencies and leaves a narrow area around cutoff frequency. The narrowness is set by the Q value.

HP1 1-pole high pass (6db/octave slope)

HP1 2-pole high pass (12db/octave slope)

BR1 Band reject, also called notch. Cuts away frequencies in a narrow area around cutoff and leaves low and high frequencies.

In Use

The output jacks on the Intellijel Korgasmatron II is marked with by a square to make it easier to find them. Output A and B are marked by black squares and the Mix output is marked by a blue square.

Korgasmatron II is an excellent filter for distortion combined with colorful filtering duties. 1V/Octave tracking makes it possible to use the Korgasmatron II as a dual sine-oscillator when self oscillating. However, just like the hardware, tracking stretches more or less 1 and half octave which limits its scope as oscillator. The Korgasmatron II is tracking best when Q knob is set fully clockwise

The highly flexible **Xfade** in combination with different filter modes makes it a easy to create moving and interesting filtering.

Intellijel µFold II

Collaborated design with David Dixon, modeled after the some of the most beloved timbre shaping modules in classic synth modules; both in terms of how the were designed electronically as well as how they sound. The µFold II is one big sweet-spot for shaping your oscillator or feedback loop for filtering.

Parameters

Stages This switch selects the number of stages of folding used, 2, 4 or 6.

Folds This knob controls the folding offset (initial level).

[Folds CV amount] This knob controls the folding amount scaled from **Fold CV** input.

Symmetry This knob sets the through-zero symmetry offset.

[Symmetry CV amount] This knob controls the symmetry amount scaled from **Sym CV** input.

Inputs

Fold CV input for controlling folding, closely linked with the **Fold CV Amount** knob (see above).

Sym CV input for controlling **Symmetry**, closely linked with the **Symmetry CV Amount** knob (see above).

In This is the audio or CV input for the µFold II. Insert whatever signal you want to fold here.

Outputs

Out This is the output of the µFold II.

In Use

Connect your oscillator to the µFold II input jack and out comes a folded signal. The µFold II also works nice to fold low-frequency signals such as LFOs. A neat trick is to connect input to output to create a white noise source with full range.

Credits

Oscar Öberg – concept, modeling, sound design. **Kristofer Ulfves** – concept, marketing, presets, user's guide. **Björn Rödseth, Kim Larsson** – modeling. **Arvid Rosén** – modeling and validation. **Patrik Holmström** – graphics programming. **Niklas Odelholm** – graphic design, programming, model validation. **Paul Shyrinskykh** – quality assurance. **Torsten Gatu** – programming. **Henrik Andersson Vogel** – marketing. **Danjel van Tijn** – hardware design and model validation. **David Dixon** – hardware design.

30 Kerry King Marshall Signature Amp

Kerry King has used Marshall amplifiers throughout his hugely successful career.

– My sound has always been more about punch than distortion. It's also creating a tonality that makes a riff sound great, so every single note and chord gets its point across without being disguised by too much gain. This amp does that in brutal fashion!

The Marshall 2203KK Kerry King Signature Series amplifier is based on the original Marshall JCM800 2203 amplifier – "The Beast" – that Kerry King used in conjunction with a graphic EQ throughout his early career to boost the signal even further. The 2203KK delivers the ferocity and tone that Kerry King's sound demands.

In this plugin, Softube has carefully modeled the 2203KK amp for you.

The amplifier

In Kerry King's own words:

– I have this one JCM800 amp that I call "The Beast", because it annihilates all others. It's as if Satan reached up, touched that head himself and conspired with Jim to create something miles ahead of the rest! It's the "Golden Child" when it comes to my sound, and it's been my main live and recording head for pretty much my entire career.

– So when it was time to work on this project, I gave Jim's R&D gurus "The Beast" plus my main graphic EQ. They did a shitload of measurements and copied both exactly – putting my unique EQ curve on a single control called "Assault" and building in an incredibly effective noise gate.

As opposed to the JCM800 2203's four EL34 power tubes, "The Beast" has four KT88s for added warmth and girth plus making it a very loud amplifier!

The speaker cabinets

Kerry King's choice of speaker is the Marshall Mode Four MF400B model, of which he uses several, both in a studio setting and as a live rig. The Marshall Mode Fours are equipped with Celestion G12K-100 speakers, giving the cabinets loads of low end to handle Kerry King's intense rhythm style playing.

Two Mode Four cabinets were carefully mic'ed and produced by world-renowned producer Terry Date and recorded at the famous Henson Studios in Los Angeles, California. The signal measurements of the provided five different setups where then carefully analyzed and recorded by Softube's Kim Larsson and Patrik Jensen.

By doing so, we have been able to carefully recreate Kerry King's sound through the production genius of Terry Date and the exquisite recording facilities of the Henson Studio.

The microphones

Different microphones do different jobs, and in this plugin we're taken advice from the experts.

Producer/engineer Terry Date has recorded Kerry King and oceans of other metal and rock acts. In the speaker section you will find five of his ready-produced setups for capturing Kerry King's sound.

FOH (PA) engineer Chris Paccou has done numerous live gigs with Kerry King, and his ready-produced live setup can be found under the second tab of the cabinet section.

The amplifier panel controls

The Cabs On switch

Where the Standby switch is usually located, there is now a switch that allows you to bypass the cab section of the plugin. This will result in only hearing the direct sound of the Kerry King Signature amplifier. This setting is useful when combining this amp with other cabinet/microphone models.

These include the Softube Bass Amp Room (including 8x10), Metal Amp Room and Vintage Amp Room (including Half Stack).

In all these plugins, the amplifier section can be bypassed. This would be the recommended setting, when combining the Kerry King Signature with the cabinets in either of those plugins.

The knobs

Presence This control operates in the power section and adds high frequencies to your sound by altering the amount of negative feedback. Turning this control clockwise adds more bite to your sound, making it crisper and more cutting.

Bass This adjusts the bottom end. Turning it clockwise increases the amount of low frequencies in the sound.

Middle This control adjusts the level of those all-important mid-range frequencies. Turning it clockwise increases the mids and fattens your sound, giving it more punch. Turning it counterclockwise reduces the mids, producing a more 'scooped' tone.

Treble Here you can adjust the top-end. Turning it clockwise increases the amount of high frequencies present in the sound, giving your guitar tone a brighter edge.

Please note: This is a modeled amplifier plugin, i.e. the controls and knobs behave as they do on the physical amp. They are not independent of each other, so consequently, altering one control can change the way the others behave. For this reason, experimentation is recommended.

Master Volume This controls the overall output level of your 2203KK. Turning it clockwise increases the volume.

Pre-Amp Volume Turning this control clockwise overdrives the pre-amp valves and creates a harmonically rich distortion.

The Beast section

"The Beast" switch This switch engages the Kerry King Signature tone. It engages the Assault and Gate circuits. Its LED will light in red when the expander circuitry is in noise reduction mode (gate closed) and green when the expander is open, letting the signal pass without any attenuation.

Gate Threshold Here you control the 2203KK internal expander circuitry. This circuitry replaces Kerry King's noise gate with an ultra-fast studio quality noise reducer unit. Instead of abruptly cutting the sound once the threshold is reached like a noise gate, the expander progressively attenuates the signal following its dynamics.

The **Threshold** control has a program dependent decay ratio allowing very slow decay times for slowly moving guitar signals thus maintaining the natural sustain and very fast decays for staccato rhythms and short pauses.

Adjust the **Threshold** knob to control the point where the noise reduction begins to work. Make sure that when adjusting the **Threshold** knob, you are not picking up hum from your equipment, as it may result in a setting higher than needed.

A setting to the right has almost no effect at all and turning the knob counterclockwise makes the **Gate** more effective.

Assault Intensity This circuitry replaces the old graphic equalizer with an electronic circuit tuned for Kerry's preferred frequency response together with user adjustable gain. Adjust the Intensity knob for the desired amount of gain boost.

Tip: Some guitar players, like Kerry King, adjust the Threshold knob to silence the amplifier with the guitar volume on full whilst others prefer to silence the amplifier with the guitar volume turned down. Feel free to experiment with both settings.

The Kerry King Cabs panels

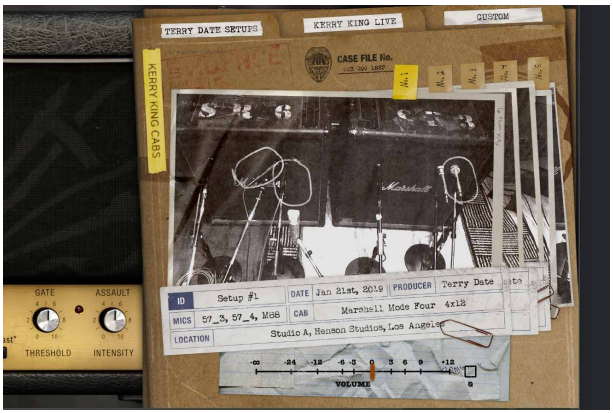
Clicking on the Kerry King Cabs sticker on the right side opens up a set of panels. Here you can tweak the microphone setup to your heart's content – mic selection, level, panning, and more.

Terry Date setups

Renowned engineer/producer Terry Date has analyzed and recorded impulses from two selected Marshall cabinets recorded with a bunch of fine microphones. The location was the legendary Henson Recording Studios in Hollywood, Los Angeles.

For this job, he chose a combination of Shure SM57, beyerdynamic M88, AKG D19 (all dynamics), and a Neumann U87 condenser microphone.

The 5 tabs give you different combinations of these mics, each excellent in their own ways.



Kerry King Live

This is the way FOH engineer Chris Paccou mics up and produces Kerry King's cabinets live. The impulses were recorded at Hovet in Stockholm, Sweden on December 5th 2018.

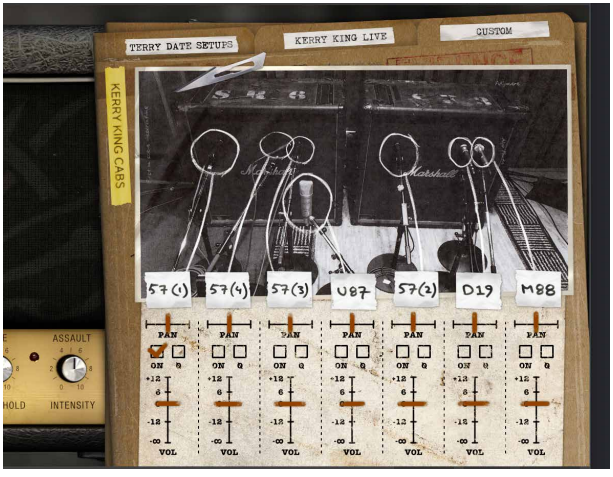
Chris Paccou used his selection of Heil PR 40 and Heil PR 30B dynamic mics on these cabs

Custom

The third tab in the Cabs section is called Custom. Here you can tweak your own sound from the variety of Terry Date's ready-produced mic setups. A total of 7 microphones can be mixed any way you want, as they have one mixer channel each.

The **Pan** slider lets you position a mic in the stereo panorama and the **Volume** slider sets the volume. A convenient **On** switch makes it easy to try out different mic combinations and the **Phase invert** switch adds further variations to the sound.

A word of warning: When using two or more microphones and pressing the Phase switch on one channel, please make sure you test the sound in mono. Two similar signals will often cancel each other out, if the phase is reversed on one channel. The result will be a thinner sound at a considerably lower volume.



Menu row

At the bottom of the plugin interface, you will see a thin black row with some buttons.

Softube logo Click here to see the version number.

Presets This button takes you to the **Preset Collection**. This is the home of loading and saving presets.

Setup You can tweak some basic settings here. Warning when deleting and/or overwriting presets can prevent you from making silly mistakes. Switching off OpenGL can solve some graphics issues. Show Value Display turns the value display at the bottom of the plugin window on and off.

Manual Link to what you're reading now.

Preset Collection

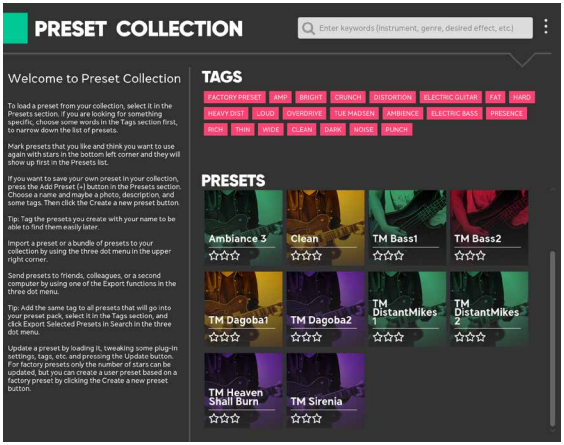
The **Preset Collection** can speed up your work considerably. Please check some presets and let them inspire you.

You can save the sounds you create by following the instructions found in **Preset Collection**. Careful tagging of presets will make life easier for you in the long run.

One difficulty with the presets for amplifier plugins in general, is that it's impossible to foresee what input gain the user sends into the plugin. Thus, a "clean" preset might sound very distorted in your setup, or maybe the crunchy presets are just too clean. A humbucker might distort the amp, while a weak single-coil pickup barely bothers the amp.

If that is the case, adjust either the volume control of your guitar or guitar interface, or the volume controls in the plugin to get the desired sound.

A common way of using Marshall amps is to keep the volume knobs at around 12 o'clock and changing the gain with the volume control of the guitar.

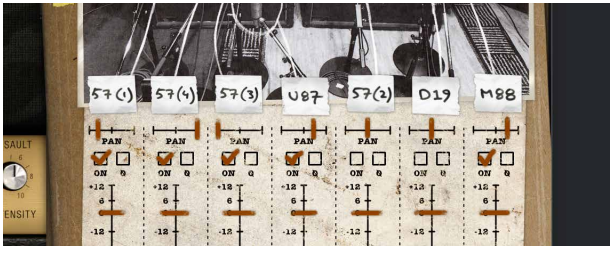

PRESET COLLECTION
Enter keywords (instrument, genre, desired effect, etc.)
Welcome to Preset Collection
To load a preset from your collection, select it in the Presets section. If you are looking for something specific, choose some words in the Tags section first, to narrow down the list of presets.
Mark presets that you like and think you want to use again with stars in the bottom left corner and they will show up first in the Presets list.
If you want to save your own preset in your collection, press the Add Preset (+) button in the Presets section. Choose a name and maybe a photo, description, and some tags. Then click the Create a new preset button.
Tip: Tag the presets you create with your name to be able to find them easily later.
Import a preset or a bundle of presets to your collection by using the three dot menu in the upper right corner.
Send presets to friends, colleagues, or a second computer by using one of the Export functions in the three dot menu.
Tip: Add the same tag to all presets that will go into your preset pack, select it in the Tags section, and click Export Selected Presets in Search in the three dot menu.
Update a preset by loading it, tweaking some plug-in settings, tags, etc. and pressing the Update button. For factory presets only the number of stars can be updated, but you can create a user preset based on a factory preset by clicking the Create a new preset button.
TAGS
FACTORY PRESET
AMP
BRIGHT
CRUNCH
DISTORTION
ELECTRIC GUITAR
FAT
HARD
HEAVY DIST
LOUD
OVERDRIVE
TUE MADSEN
AMBIENCE
ELECTRIC BASS
PRESENCE
RICH
THIN
WIDE
CLEAN
DARK
NOISE
PUNCH
PRESETS
Ambiance 3
Clean
TM Bass1
TM Bass2
TM Dagoba1
TM Dagoba2
TM DistantMikes 1
TM DistantMikes 2
TM Heaven Shall Burn
TM Sirenia

Mono and stereo operation

The Kerry King Signature operates, as nearly all other Softube plugins, in mono and stereo. Mono in → mono out. Stereo in → stereo out.

This plugin can also turn a mono signal into a stereophonic sound; the microphone channels in the Custom tab has Pan sliders, that let you freely adjust the stereo width.

> Please note, that some digital audio workstations (DAWs) require you to set a track to stereo to actually output a stereo sound.



Keyboard

Holding down a key on your computer keyboard will add some extra functionality.

(Mac) or Ctrl (Windows)	This key gives you fine adjustment when tweaking a knob.
Alt	Reset to default value.

Credits

Patrik Jensen – project management. **Kim Larsson** and **Mattias Olsson** – DSP modeling. **Patrik Holmström** and **Pelle Serander** – framework programming. **Kerry King** – sound specifications. **Terry Date** – studio sound design and presets. **Chris Paccou** – live sound design and presets. **Niklas Odelholm** – general graphic design. **Ulf Ekelöf** – graphic design amplifier. **Andreas Pettersson** – graphic design cabs. **Nis Wegmann** – additional graphic design. **Emanuel Enbäre** – quality assurance. **Sven Bornemark** – manual copywrite. **Fanny Hökars** – manual design.

31 Marshall Plexi Super Lead 1959

It is impossible to exaggerate the iconic status of the Marshall Plexi Super Lead 1959 amplifier. Jimi Hendrix, Pete Townshend, Eric Clapton, Angus Young, Eddie Van Halen, Yngwie Malmsteen – the list of Plexi users is a veritable who's who of rock 'n' roll history. Forget all stereotypes about British reserved manners and stiff upper lips. The Plexi is brash, rude, and gives you the instantly recognizable sound of rock 'n' roll rebellion.

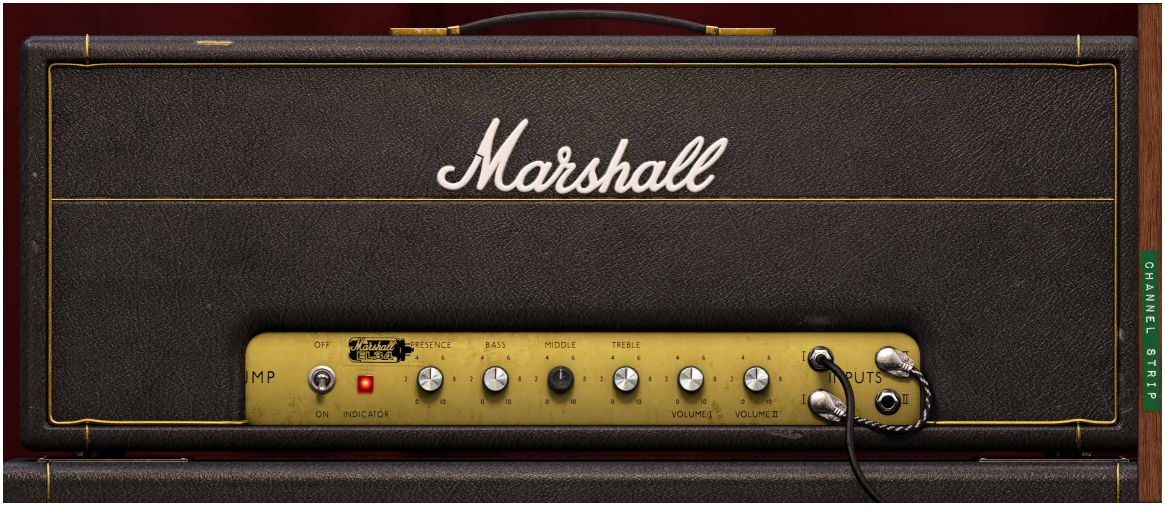
This plugin is a "what you see is what you get" product. You should be able to intuitively learn to use it within minutes, so that you can work fast and efficiently with it.

Softube has worked closely with Marshall's product experts and legendary recording engineer Tony Platt – who engineered AC/DC's Highway to Hell and Back in Black – to ensure that the Plexi was captured in the best possible manner. The result is this plugin that allow computer-based musicians to enjoy a legendary Marshall stack without getting evicted.

The amplifier

The particular amplifier that Softube modeled is not just any Marshall Plexi. In the making of this plugin, Softube worked closely with Marshall's product expert Chris George, who lent us this amp from Marshall's own reference collection.

This very amp, built in 1967 and kept in pristine shape (at least on the inside) by Marshall's technical staff, is Marshall's own reference to how a Plexi should sound.



The speaker cabinet

The goal for this plugin was to capture how the Plexi sounded when it was new back in 1967 – to get the sound Jim Marshall himself intended.

To capture the typical late sixties tone, Tony and Chris – after much testing at Marshall headquarters – chose to use a straight 1960BHW 4x12 inch speaker cabinet, loaded with Celestion G12H-30's. These are relatively low wattage speakers, similar to those used at the time, which will saturate more easily compared to modern constructions. This adds a lot of character to the end result.

The particular speakers in this cabinet were not from the sixties. Speakers that have spent nearly 50 years reproducing the sound of 100 watt guitar amplifiers, will have deteriorated and give a much different sound compared to what was intended. The speakers used had however been well broken in to reach their full potential and liveliness.

The microphones

Tony chose to make three different setups with three microphones in each – two close mics and one room mic.

Clicking on the **Channel Strip** sticker on the right side, gives you access the control panel. Here you can select between the three sets of microphones and adjust their individual levels and panning. The settings are named **FET**, **Valve** and **Dynamic** to indicate what microphone types were used.

The Amplifier panel controls

The knobs

Presence	Increases the amount of presence – a treble boost accomplished by reducing the amount of high frequencies being subjected to negative feedback in the power amp section.
Bass, Middle, and Treble	These tone controls determine the frequency content from the amplifier.
Volume I	Controls the volume of the brighter sounding Input I.
Volume II	Controls the volume of the darker sounding Input II.

Please note: Clicking above the tone and volume controls resets the corresponding knob to its 12 o'clock position.

The Input configurations

The inputs

The two **Input I** sockets provide a bright sound, and the **Input II** sockets have a darker sound.

The lower two sockets have a lower gain (–6 decibel) than the upper inputs.

Patch switching

There are numerous combinations of the high and low gain inputs for **Input I** and **Input II**. You can access those variations by clicking on the input sockets.

Clicking one of the four inputs selects it as the main input from the guitar (black cable). Clicking the selected guitar input again rotates between the available patching patterns (with the short black/white fabric-covered patch cable).

By patching the inputs, you will get different input gain and different tonality from the amp. **Input II** is very dark sounding, and by combining it with **Input I** you can work the **Volume** knobs almost like an EQ.

The most common way of patching is going into **Input I** at the top, and patching from the low **Input I** to the top **Input II**, but feel free to experiment!

The Channel Strip panel

Clicking on the Channel Strip sticker on the right side takes you to a set of controls. Here you can fine-tune microphone selection, level, panning, and more.

Equaliser and Volume Main Out

Low Neutral/bypass in the 12 o'clock setting. Turning clockwise boosts the low end with a shelving filter. Turning counterclockwise cuts the low end with a low cut (high pass) filter.

High Neutral/bypass in the 12 o'clock position. Turning the knob clockwise boosts the treble with a shelving filter. Turning it counterclockwise cuts the treble, also with a shelving filter.

Volume Adjusts the output v entire plug-in.

Cabinet Microphone Select

Here you can select which set of microphones to use. See description in the next section.

Microphone mixer

Solo	Press this button to listen to only one microphone. Hold down Shift to select several mics.
Pan	Places the microphone channel in the desired position in the stereo panorama.
Channel faders	Adjust the volume of the respective microphone channels.

Cabinet and microphones

The 16 ohm Marshall 1960 BHW cabinet was recorded with a legendary range of microphones, all carefully selected to give a wide variety of tones, and carefully positioned to sound fantastic when used in combination with each other.

Valve microphones

A thick and solid sound, with a slightly rounded top.

Mic 1	Neumann U 67 (cardioid, –10 dB pad activated)
Mic 2	Neumann U 67 (cardioid, –10 dB pad activated)
Room mic	AKG C12

FET microphones

A slightly more open sound with lots of ambience from the room mic.

Mic 1	Josephson E22
Mic 2	Neumann U 87 (cardioid, –10dB pad activated)
Room mic	Coles 4038

Dynamic microphones

Typical dynamic sound with a solid low mid-range from the 57 and aggressive top from the 609.

Mic 1	Shure SM57
Mic 2	Sennheiser E609
Room mic	Neumann U 87 (omni)

Off (no cabinet or microphone)

In this position, the cabinet and microphone emulations are bypassed, so that only the direct sound from the Marshall Plexi amplifier head is heard. This setting is useful when combining this amp with other cabinet/microphone models.

These include the Softube Bass Amp Room (including 8x10), Metal Amp Room and Vintage Amp Room (including Half Stack).

In all these plugins, the amplifier section can be bypassed. This would be the recommended setting, when combining the Marshall Plexi Super Lead 1959 with the cabinets in either of those plugins.

Menu row

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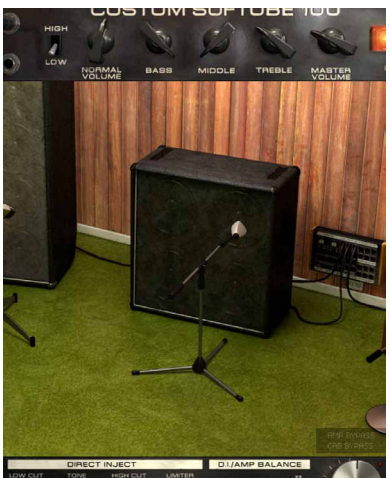
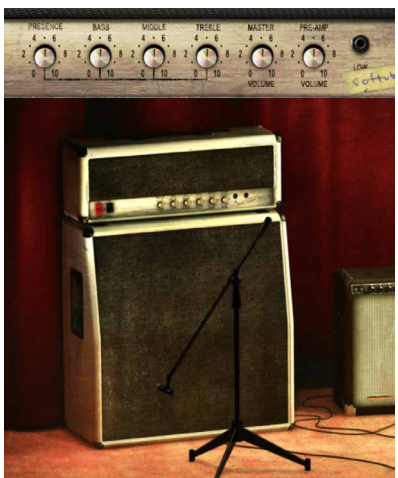
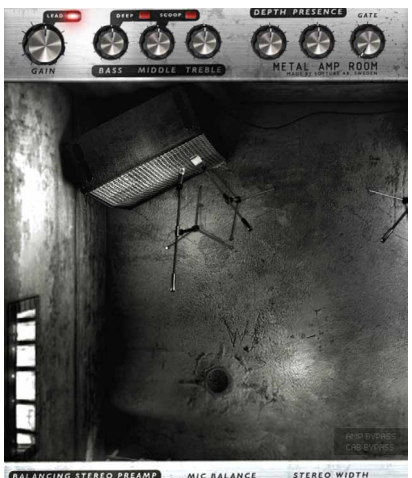
Manual	Link to what you're reading now.

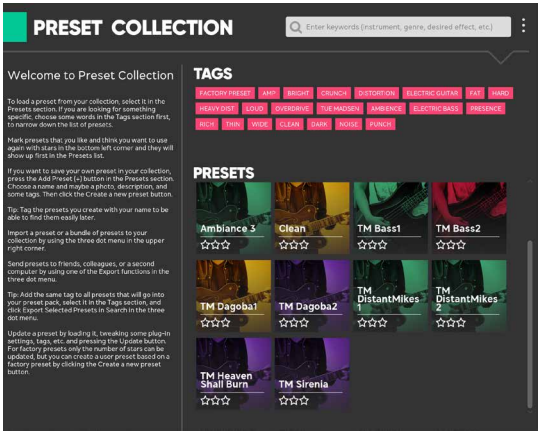






Preset Collection

The **Preset Collection** can speed up your work considerably. Please check some presets and let them inspire you.



You can save the sounds you create by following the instructions found in **Preset Collection**. Careful tagging of presets will make life easier for you in the long run.

One difficulty with the presets for amplifier plugins in general, is that it's impossible to foresee what input gain the user sends into the plugin. Thus, a "clean" preset might sound very distorted in your setup, or maybe the crunchy presets are just too clean. A humbucker might distort the amp, while a weak single-coil pickup barely bothers the amp.

If that is the case, adjust either the volume control of your guitar or guitar interface, or the volume controls in the plugin to get the desired sound.

A common way of using Marshall amps is to keep the volume knobs at around 12 o'clock and changing the gain with the volume control of the guitar.

Mono and stereo operation

The Marshall Plexi Super Lead 1959 operates, as nearly all other Softube plugins, in mono and stereo. Mono in → mono out. Stereo in → stereo out.

This plugin can also turn a mono signal into a stereophonic sound; the microphone channels in the **Channel Strip** have **Pan** knobs that let you freely adjust the stereo width.

> Please note, that some digital audio workstations (DAWs) require you to set a track to stereo to actually output a stereo sound.

Keyboard

Holding down a key on your computer keyboard will add some extra functionality.

⌘(Mac) or Ctrl (Windows)	This key gives you fine adjustment when tweaking a knob.
Alt	Reset to default value.
Shift	Solo several mics in the **Channel Strip.**

Credits

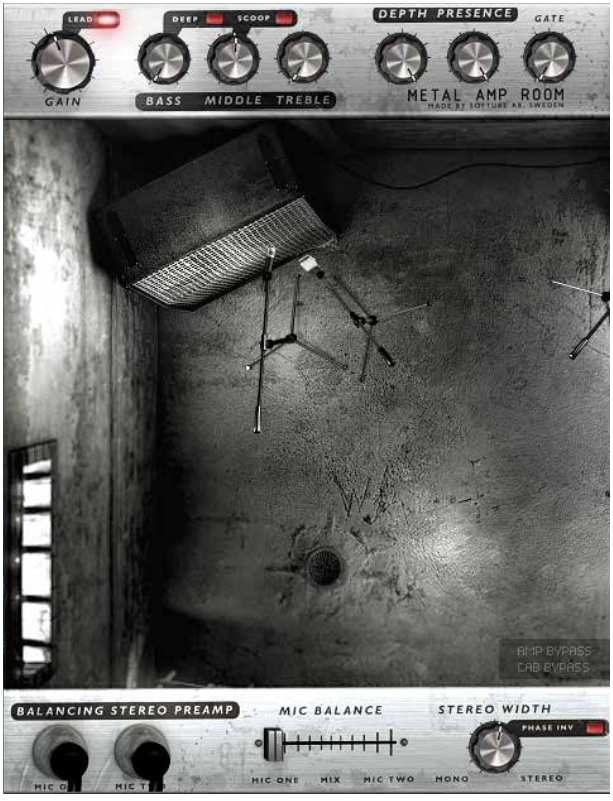
Patrik Jensen – project management. **Henrik Andersson Vogel** – project management, manual and marketing. **Niklas Odelholm** – product design, cabinet modeling, graphic design, presets. **Arvid Rosén** – amp modeling. **Oscar Öberg** – power amp modeling. **Kim Larsson** and **Mattias Olsson** – additional DSP modeling. **Tony Platt** – sound design and presets. **Chris George** – Marshall expert, sound design and presets. **Patrik Holmström** and **Torsten Gatu** – framework programming. **Pelle Serander** – framework. **Ulf Ekelöf** – 3D graphics. **Erik Hampusgård** – configurations and building. **Paul Shyrinskykh** and **Emanuel Enbäre** – quality assurance. **Sven Bornemark** – manual copywrite. **Fanny Hökars** – manual design.

32 Metal Amp Room

Introduction

METAL AMP ROOM IS JUST WHAT IT SAYS IT IS. It's an amp room, just like the other Softube plug-ins Vintage and Bass Amp Room, but it's designed for Metal. It consists of three parts, of which the first part is the amp itself. It was carefully selected to have a aggressive sound with a smooth high end and never ending sustain, and special care was taken to make sure that the palm mute playing style hits you right in the stomach. We also added a lower gain (not to confuse with low gain!) preamp to accommodate for a more traditional playing style.

Second, and perhaps the most important of the building blocks, are the cabinets. You can choose between two different fourbytwelve cabinets, and each cabinet has two mics. A dynamic microphone for the classic metal sound, and a condenser mic with a fuller and richer sound. Just playing around with different cabinets and mic settings is a science in itself, and we have worked really hard to make it as easy as possible for you to find the sound you look for. To make this even easier, we had to design the third building block – the BALANCING STEREO PREAMP. This is basically two channel strips with a volume, a pan and a phase control each, that we combined into a single channel strip with a MIC BALANCE control instead of two volumes and a STEREO WIDTH control instead of two pan knobs. Just to make the handling easy, without compromising the sound.



And like our other products, such as Vintage Amp Room, Metal Amp Room has none of the added gadgets or weird-sounding presets sometimes found in simulation software. Focusing on quality and accuracy, it gives you a hard hitting metal sound that is authentically raw. The goal is not to give you a preproduced, ultimate guitar sound, but to provide you with the ultimate tool to create your own. Because after all, only you know exactly what sound you want.

Product Philosophy

Everything about this product is authentic. The raw and dynamic sound is an absolute replica of what you would get if you had this amp and these cabinets set up in a real studio. What you see on the screen are photorealistic 3D renderings of the complete setups, and you get to move actual mics around when deciding which positions are best, from near field offaxis to far field and back, continuously, without preset positions.

The idea is that using Metal Amp Room should be like working directly with this amplifier set up in a real studio, with two microphones in front of each cabinet, connected to your DAW. And we kept it simple. It has no knobs with dubious or unintuitive functionality, and no added gadgets or ridiculously fakesounding effects. We simply provide you with the same tools you have in a real studio, but in digital form. Then it's up to you to add compressors, EQ's, reverb, or any other effects, to get the sound you want. In short, you need the same skills to master this software as you need when dealing with a real studio setup. You can get back to doing what you do best, because music production is about using your ears, not your computer skills.

Supernormalize

The "supernormalize" feature makes sure that the output signal from Amp Room always has a reasonable volume. From a user perspective, this primarily means that the output never exceeds 0 dB and you don't have to fiddle with tons of output volume knobs. And even with low settings on the gain knob the output volume will be within a reasonable level and ready for digital recordings. Compare this to the real amp, which easily can have a dynamic range of 100 dB. From a technical perspective, this means that the signal path within Amp Room always has the highest dynamic range possible, and you won't lose any bits on the way.

User Interface

SIMPLICITY WAS THE GOAL when we developed this product–the amps and mics should work just as they do in the real studio. We have tried to imitate that as far as it is possible, and if you have experience of working in real studios, you will notice that Metal Amp Room handles and sounds just the way you expect it to.

Amp Panel

Room View

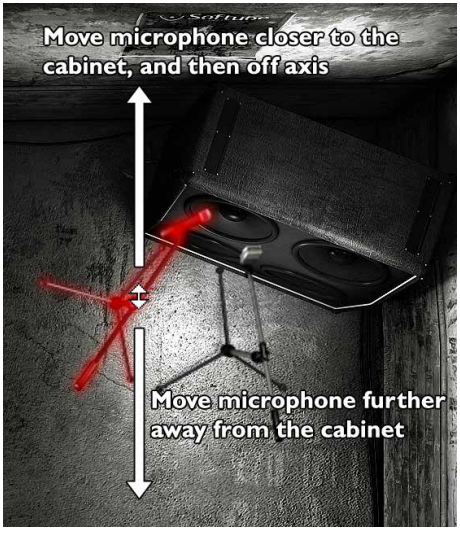
Mic Panel

Amp Panel (Top Area)

In the amp panel you can alter the settings of the amp by clicking the knob and dragging the mouse up and down or left to right. Switches will be switched either by clicking on them, or by click-and-dragging the mouse. In some hosts you can change the behaviour of the mouse, but the default behaviour of Metal Amp Room is the one described above.

Room View (Middle Area)

In the room view you can do two things, select the other cabinet or change the mic's positions.



Mic Position

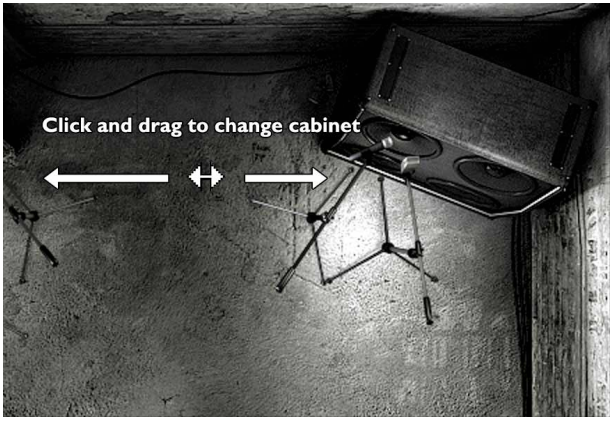
When the mouse is located over the mic stand, the mouse pointe changes to an "up-down" arrow and the mic gets a red light saber glow. Click and drag the mouse up or down to change the position of the mic stand. The microphone moves along a predetermined path, so you only need to move the mouse up or down, just as if you were changing a knob.

Linked Mics

If you press and hold Shift while moving a mic, the other mic will also move to the same position respectively.

Cabinet Selection

You can change the cabinet by clicking on the background and dragging the mouse to the left or right. The mouse pointer becomes a "left-right" arrow when you are pointing at the background to indicate that it is possible to change cabinet.



Mic Panel (Bottom Area)

In the mic panel you can change the balance and volume of the two microphones by setting the level from the two mics with the fader or the using the knob to alter the stereo width. The mic panel follows the cabinet selection so that you can have different settings for the different cabinets.

> Clicking on the background while holding the Shift-key toggles through the cabs without any sliding animations.

1. Lead/Rhythm channel select
2. Deep bass
3. Scoop mids
4. Gate
5. Gain
6. Bass
7. Middle
8. Treble
9. Depth
10. Presence

The Amplifier

THE AMPLIFIER SIMULATED IN THIS PLUG-IN is actually a hybrid between two different amps, where the LEAD channel (**Lead** = ON) represents one whole amp, and the RHYTHM channel (**Lead** = OFF) is a simulation of the same amp but with the input stage taken from an amplifier with a lower gain.

The LEAD channel is the main channel – the one to go to – and extreme care has been taken to make sure that the palm muted guitar hits you right in your face.

The RHYTHM channel represents a more classic rock sound, and is perfect for chords containing more than the root and the fifth...

We also took the liberty to add our own noise gate to the amplifier. It was specifically developed for guitar and focus on an extremely fast response. You will notice that the gate closes (turns off the volume) more slowly if you let your notes decay slowly than if you mute your notes. So you can both play fast riffs and have the gate kick in between the chords, and play slow solos with a not so aggressive gate – all on the same setting.

Gain This knob controls the amount of distortion. When you have selected the RHYTHM channel, the **Gain** knob can be used to adjust the amount of preamp distortion. When you have selected the LEAD channel, the **Gain knob is** mostly used to shape the character of the distortion. When the **Gain** knob is set at 9 o'clock, you'll get a tight distortion with a fast response. As you increase the **Gain** you will get more distortion, but it won't be as tight as in the lower settings. A reasonable setting is around 12 o'clock.

Lead This switch selects between the RHYTHM channel (light is off) and the LEAD channel (light is lit).

Bass, Middle, and Treble	These knobs are the tone controls of the amplifier. This circuit is located after the preamp distortion and is used to shape the sound of the preamp distortion.
Deep	Boosts the ultra low frequencies.
Scoop	Scoops out the mid to create a tighter sound.
Depth, Presence	These are used to control the amount of low and high frequencies in the feedback loop of the power amp and thus the amount of bass and treble in the power amp. Turn them up to get a low or a high frequency boost.
Gate	A noise gate specifically designed for a guitar input. Mute your guitar and turn up the knob until the noise disappears.

The **Gate** is program-dependent, which means that if you play fast and tight riffs and end your notes distinctly, you will get a very fast gate. If you on the other hand play sustained notes, the gate will close more gently.

The Cabinets

The Metal Cabinet

The left cabinet is called the METAL CABINET since it has a stylish metal grille and an uncompromising metal sound. This is a predictable cabinet in the sense that there are very few surprises as you move the mics around. The METAL CABINET will give you an emphasis on mid-range frequencies, which makes it excellent for modern, tight styles. Very little post-processing (in the form of EQs, etc.) is usually needed since the raw sound has a produced character. Just do some minor tweaks to make it fit in the mix and you will be fine.

The Black Cabinet

To the right is the BLACK CABINET, which has a classic rock 4x12 sound. With this one, there are plenty of bass resonances as you move the mics around, producing a lively, sometimes almost unruly sound. To get a vintage sound, set the balance all the way to MIC ONE or blend in MIC TWO to get a more scooped sounds. Depending on how the rest of the mix sounds and the exact positions of the mics, the BLACK CABINET could need a bit of work with the amp's tone stack and perhaps some external EQing to blend in properly.

1. Mic balance
2. Stereo width
3. Phase invert

The Microphones

Mic One

MIC ONE (to the left in both cabinet views) is a classic dynamic microphone. It has a tight mid range that often makes out the core of the sound. The character is focused and controlled when the mic is close to the cabinet, and gets a thinner ambient room quality (METAL CABINET) or more boomy and unpredictable (BLACK CABINET) as it is moved back. This mic is very directional, so when it is between the close onaxis and offaxis positions, small changes in angle will have a big effect on the sound.

Mic Two

MIC TWO (to the right) is a condenser microphone. Use it to add some scooped character to the mid range of MIC ONE or use it alone for a more open, broader, high bandwidth sound. Just like MIC ONE, the character is focused when the mic is close to the cabinet and gets more room ambiance (METAL CABINET) or more boomy (BLACK CABINET) as it is moved back. The far-field sound is still more controlled than the far-field sound of MIC ONE though.

The Balancing Stereo Preamps

EACH CABINET HAS THE "Balancing Stereo Preamp". It isn't modeled after some particular piece of gear, but was instead designed to make it as easy as possible to tailor the sound from the two microphones. The main ideas are:

1. It should be ridiculously easy to use without having to worry about anything but the sound you want to get.
2. It should be easy to monitor each mic (which is done by moving the **Mic Balance** fader fully to each side).
3. It should be easy to blend the mics and still get a stereo image that is balanced to the center.

Mic Balance	Fades between the two mics. If it's fully to the left you'll get the sound from the left mic, and if it's set to the right you'll get the sound from the rightmost mic. If the **Stereo Width** knob is set at MONO, this is exactly like an ordinary cross fader. Set it as in the picture above, and you'll get about 75% of the left mic (MIC ONE) and 25% of the right mic (MIC TWO).
Stereo Width	This knob alters the stereo width of the signal.
Phase Invert	This switch inverts the phase of MIC TWO.

Example Settings

One Mic

Mic Balance=0 (LEFT)
Stereo Width=anything

Only the left mic is selected, and since we only have one source (the left mic) the **Stereo Width** knob won't do anything and we will have the same output in both the left and right channel.

Both Mics, Mono Output

Mic Balance=0.5 (MIDDLE)
Stereo Width=0 (MONO)

With the **Mic Balance** in the middle, we have an equal amount of both mics, and since the **Stereo Width** is MONO, both the left and the right channel will sound the same. Both channels will have 50% of the left mic and 50% of the right mic.

Full Separation

Mic Balance=0.5 (MIDDLE)
Stereo Width=1 (STEREO)

We're in full stereo and equal amount of both mics. The left mic (MIC ONE) will only be heard in the left channel, and the right mic (MIC TWO) in the right channel.

Half and Half

Mic Balance = 0.5 (MIDDLE)
Stereo Width = 0.5 (halfway between STEREO and MONO)

Equal amount of both mics, but since the **Stereo Width** control isn't fully in stereo, we will have some bleed between the two channels. The left channel will contain about 75% of the left mic and 25% of the right mic, and vice versa with the right channel.

Mono and Stereo Operation

Metal Amp Room is designed to be a MONO-TO-STEREO plug-in. One guitar input, two mic outputs.

For best results, and if possible, insert the plug-in as MONO-TO-STEREO or a STEREO. If it has been inserted as a stereo plug-in, it will sum the two inputs (left and right) before processing the audio.

Using Pan and Balance

Be careful when you use the balance control of the stereo track. If you for instance have a great stereo sound using both the left and right mic, and then balances the track using the balance control in your host, you will change the sound of the output. For instance, if you want the output from Metal Amp Room to be fully panned to the right, set the **Stereo Width** control to MONO before you balance it to a side. Rule of thumb is that if your balance the stereo track halfway to one side, the **Stereo Width** control should be set halfway between MONO and STEREO.

> If you balance the stereo output using the mixer in your DAW, some stereo information will get lost. Use the **Stereo Width** control to compensate for that. If you balance the track fully towards one side, set **Stereo Width** = MONO. If you balance the track halfway towards one side, set **Stereo Width** half-way between STEREO and MONO.

Bypassing Amps or Cabs

You can choose to bypass the amp or the cabinet by selecting AMP BYPASS or CAB BYPASS from the small box in the lower right corner.

This is very useful if you want to use Metal Amp Room with an external speaker (or speaker plugin) or if you just want to add a cabinet simulation to a track, such as a recorded line out from your amplifier.

> By bypassing the cabinets in Metal Amp Room, you can use the cabinets from the other Amp Room plugins. Just insert, for example, Bass Amp Room (with bypassed amplifier) on the track after Metal Amp Room (with a bypassed cabinet).

Buying Recommendations

If you like the sound from Metal Amp Room and would like to get that sound using the real deal (let's face it, a real amp is always sexier than a plugin), here are some buying recommendations.

Amplifier

The sound of the amplifier is very much inspired by the one and only metal amp, the Engl Powerball. To get a nice rhythm channel, we utilized our soldering gear and bred our own JCM800/Powerball mutant.

Metal Cabinet

If the Metal Cabinet is your thing, run out and get a slanted Engl E 412 V 4x12 cabinet for live gigs.

Black Cabinet

For that classic vintage sound, we recommend a brand new straight Marshall 4x12 cabinet.

Room

The room is a very important part of the sound, and we had the opportunity of visiting The Haunted while they were laying down tracks for their new album in the In Flames studio (previously known as Studio Fredman). Tue Madsen and Patrik Jensen made sure that all the cabs and the microphones sounded just the way they should. You cannot get more metal than that.

ENGL IS A REGISTERED TRADEMARK OF BEATE AUSFLUG AND EDMUND ENGL. MARSHALL IS A REGISTERED TRADEMARK OF MARSHALL AMPLIFICATION PLC.

Gate and Preamp

The noise gate and balancing preamp was developed by Softube, and has not yet had the chance to be embodied in a real piece of gear.

Credits

Niklas Odelholm – modeling and sound design. **Oscar Öberg** – concept, preamp modeling, sound and DSP-programming. **Arvid Rosén** – power amp modeling and sound programming. **Torsten Gatu** – interface and DSP programming. **Ulf Ekelöf** – graphics programming and 3D rendering. **Patrik Jensen** – amp selection and sound design. **Tue Madsen** – sound design and cabinet measurements.

33 Modular

BEFORE THERE WERE MINIMOOGS, Odysseys or any of the Japanese compact performance synthesizers, there was the modular synthesizer. The modular synthesizer consisted of the core building blocks—modules—for sound sculpting; tone generators (also known as oscillators), fixed filters and modifiers. These were separate units which could be interconnected, usually using patch cords. The fact that the individual modules could be patched in a number of configurations opened to a wide range of sonic options.

Doepfer And The Eurorack Synth

In the mid 1990s, there was a modular synth revival when German synth designer **Dieter Doepfer** launched his range of modular synthesizers. Doepfer's system was called A-100, and the modules were housed in Eurorack casings, a rack format previously used extensively in the telecommunications industry. Soon, a large number of synth vendors started coming out with their own modules for the Eurorack standard. Today, there are over 4000 Eurorack modules available from over 150 different vendors.

Softube Modular

Softube Modular is a virtual Eurorack modular synthesizer, which is used as a plug-in in any major DAW. Modular was developed in close collaboration with Dieter Doepfer, and six Doepfer modules are included in the basic Modular package, along with more than 20 of Softube's own utility modules. More modules can be purchased separately to expand the options from the basic package.

Modular can be used just like an actual Eurorack modular synth, by adding different modules and combining

them using virtual patch cords. A Eurorack modular synth is by definition monophonic and so is also the Softube Modular, although a certain degree of polyphony can be achieved. More of this later. The analog synth modules have been modeled, component by component, by Softube's engineering team. Therefore, Modular is sonically impossible to distinguish from its analog counterparts. Furthermore, just like a real life Eurorack synth, any audio signal can be fed into the modules. To facilitate using Modular this way, we have created a separate plug-in called Modular FX.

It goes without saying that the experience of connecting actual patch cords and tweaking physical synth modules can never be replicated in the software domain. But the fact that Modular resides in the computer opens to a few possibilities that are not available with physical Eurorack synthesizers. First of all, patches (complete setups of modules, their interconnections and settings) can be easily saved, instantly recalled and shared with other Modular users. Second, a large number of professionally made presets are available. That makes Modular instantly usable also to inexperienced users—and offers a great way to learn more about modular synths. Furthermore, Modular offers Performance modules, which makes it easier to highlight and adjust a select number of parameters on the fly, even with complicated patches. This makes Modular ideal to use for live performances or for inspiring music production in the studio. Plus—if you have access to physical synth modules, you can actually interconnect these with Softube Modular via an audio interface.

Quickstart Guide

If you're familiar with working with software (and hardware) synthesizers, this section may be all you need to get started. Refer to the in-depth sections of this manual to learn the details.

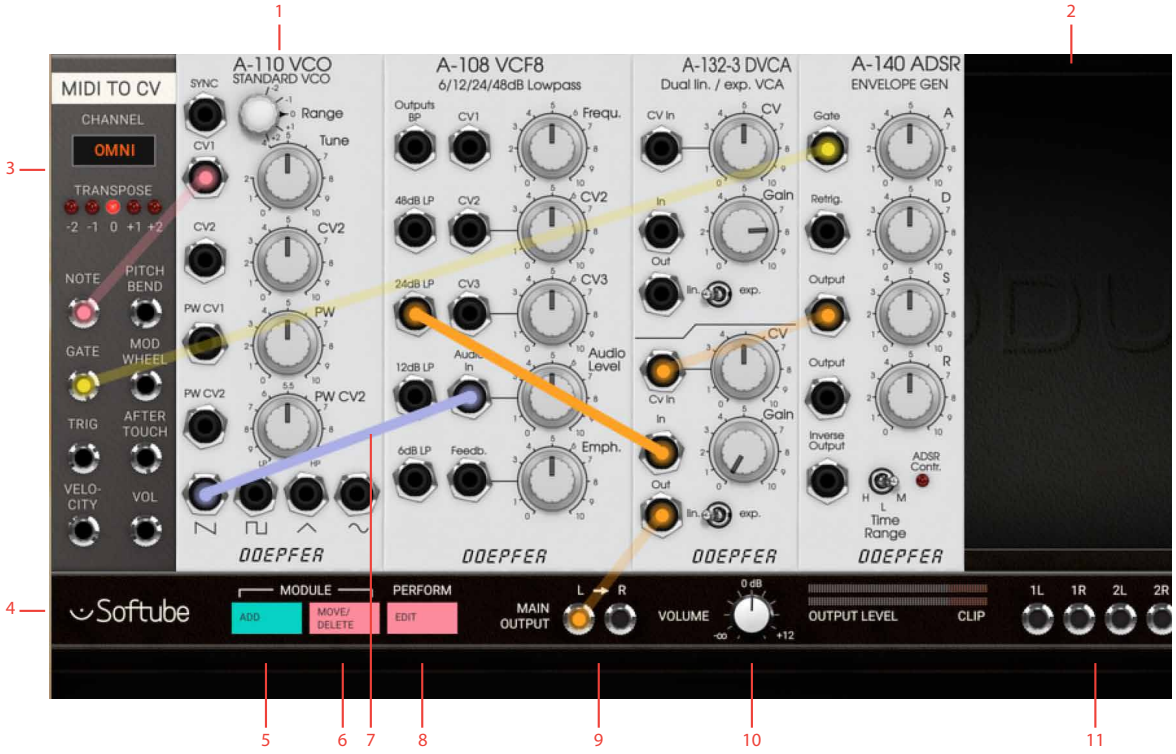
When launching Modular, you will find that its default state is an empty patch, and as no modules have been added to the rack yet, you find yourself presented a choice of modules in the MODULE SELECT VIEW. By selecting modules you can start to create your patch. You can also open a preset patch from the extensive factory library in your DAW. This is slightly different depending on which DAW and plug-in format you're using (please refer to your DAW's User Manual). If you open a preset patch, you will note that the patch cords appear when you move the mouse pointer over them, and otherwise fade away.

> To add multiple modules without exiting the Module Select view, press Shift while clicking to select.

To connect two modules, click and hold a jack. To facilitate patching, the inputs to which the patch cord can be connected will be marked with a color, green (INPUTS) and red (OUTPUTS). An output can only be connected to an input and vice versa. Simply drag and drop the cord on the desired jack. Multiple patch cords can be routed from an output (red) to several destination inputs (green), but an input jack can only accept one patch cord at the time. If you add more modules than fits your screen, you can scroll to see them all by clicking and dragging anywhere, except over a knob or jack. Scrolling can also be performed by smart gestures or scroll wheel if your computer supports this. Any patch can contain up to 100 modules, depending on the size of the modules. Just like a physical Eurorack synth, to hear your creation, you will need to finish your patch by connecting it to Modular's outputs. These are located in the CENTER ROW. Similarly, in order for your patch to be controllable via MIDI, you need to add the **MIDI to CV** module to interface with your DAW. Getting MIDI clocks from your DAW is achieved through the **DAW Sync** module.

User Interface

These are the main sections and features of Modular.



1. This is a Softube emulation of a Doepfer A-110 Oscillator module.
2. This is the empty rack space where new modules are placed.
3. To make Modular respond to incoming MIDI signals from your DAW or a MIDI controller, you need a MIDI To CV module in your patch.
4. Center Row. Here, you have access to Modular´s main functions, as well as inputs and outputs.
5. Click to enter Module Select View, where you select which modules to use in your patch.
6. Enter and exit Move and Delete mode.
7. This patch cord connects one of the oscillator´s outputs to the input of the Doepfer Filter module.
8. Enter Performance knobs linking mode.
9. Main Output. Both these need to be connected to achieve a stereo image to your DAW. Left output is normalized to the right, so connecting only L will result in a mono signal output on both channels.
10. Output volume and meter.
11. Aux Outputs.

Overview

The Softube Modular plugin consists of an empty Eurorack that can be filled with Eurorack modules, your building blocks, for building your patch. The CENTER ROW is situated between the first and second row of your rack area. The CENTER ROW consists of the buttons for **adding modules, moving around or deleting** them, as well as the button for **editing your Performance modules**.

Also situated on the CENTER ROW is the INPUT (only on Modular FX) and OUTPUT jacks for audio. Here's also the main **Output Volume** knob and its **Output Level** meter which indicates if the signal is clipping. The Modular Eurorack workspace can be scrolled up and down by clicking and dragging on the background or on the module panels in the patch view. Using the scroll wheel on your mouse or smart gestures (two finger scrolling) is also supported.

When scrolling, the CENTER ROW will scroll along, but stay at the top of the screen. Controls within modules (knobs, slider, switches) can be moved by clicking and dragging the mouse cursor over the parameter. A parameter can be reset to its pre-set value by clicking while holding ALT key. Fine adjustments can be done by holding ⌘ (Mac) or CTRL (Win), while clicking and dragging on a parameter.

Handling Modules

The modules are the building blocks from which you create your modular synth patch, just like in a real life Eurorack modular. When you launch Modular, you will see the two top rows of rack space (empty by default), divided by the horizontal CENTER ROW. The rack space is where the modules will go as you add them. You can scroll down to see additional empty rows of rack space.

On the far left side of the CENTER ROW, two buttons are labeled **Module**, one to **Add** and the other to **Move/Delete**. Clicking **Add** opens the MODULE SELECT VIEW. Here, you will see all your available modules, sorted by brand and type. Placing your mouse pointer over a module opens a brief description. The details of each module is covered in "Modules In Detail" on page 286. Clicking a module adds it to the rack. You can hold SHIFT and click to add several modules at once to the rack. Exit MODULE SELECT VIEW window by clicking the pink "X" icon in the upper right corner.

Move/Delete View

Having added modules to your rack, you can click the **Move/Delete** button in the CENTER ROW to move or delete them. While in MOVE/DELETE VIEW, the **Move/Delete** button is lit and the added modules are darkened with an added "X" mark in each upper right corner. Clicking on this "X" will delete the module.

By clicking at a module, it becomes highlighted, indicating that it is now selected for moving. Click on an empty space in the rack to where you want to move the selected module or move the module by clicking and dragging the module to the desired location. While being moved, the module will turn opaque when you drop it to a new location. If you drop it while it's semi-transparent it will be positioned back to its previous position.

Exit MOVE/DELETE VIEW by clicking on **Move/ Delete** button again. The button is unlit as you're back in normal PATCH VIEW.

Patching

When you have added modules to the rack, you need to manually connect them using patch cords. To add a connection, click and hold an output jack on a module. A circle appears, and if you move the mouse (still holding the mouse button down), a patch cord is drawn from the output.

Any input jacks that can be used as a destination for your current connection will turn green. If you are instead patching from an input, all the available destinations will of course be outputs, marked red. Drag the patch cord to one of these and let go—the connec-

tion has been made. When you move away the mouse pointer, you will see the patch cord fades away. The two connected jacks will be marked with the same color as the patch cord, and if you move the mouse pointer back near any of the patched jacks, the cord will reappear.

To alter a connection, click on a jack and move the cord to another jack. Since multiple connections can be patched from the same output jack, patchcords can only be removed from the "input-jack side". To remove a connection, drag the cord on the input jack side to somewhere where there is no other jack and let go. Multiple cables can be stacked from the same output but only one can connect to the same input. This can be useful for example when using a single envelope generator output to control multiple other modules.

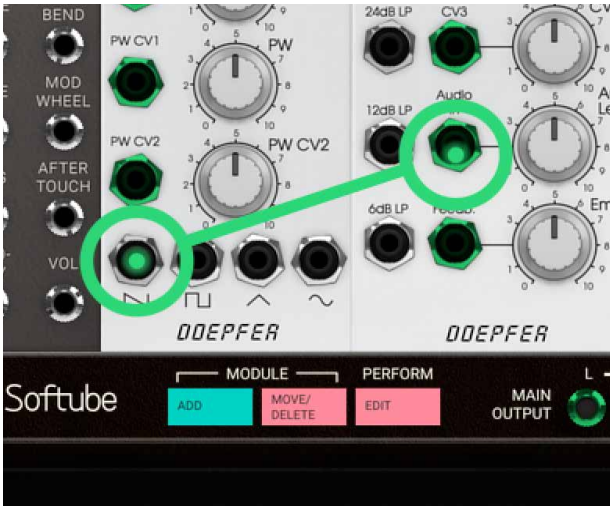
The Output Section

Any patch must end with a connection to the **Main Output** jacks on the CENTER ROW, otherwise, no sound will come out of Modular. The LEFT output is normalled to the RIGHT, which means that if your patch is only connected to the LEFT output, the sound will be in mono. To create a stereo patch, both LEFT and Right outputs must be used. The **Volume** knob lets you adjust the output volume to reasonable levels, as indicated by the **Output Level** meter. Avoid running this into the red segments, labeled CLIP.

The CENTER ROW also has four stereo pairs of auxiliary outputs, labeled **Aux Outputs**. These can be used to send separate outputs to your DAW.

> This can be useful for further processing in your DAW, or to route signals from Modular to a hardware Eurorack system.

The **Block DC On Aux Out** buttons is a feature used to make the Aux outputs sending ordinary AC (audio voltage) in order to interface with your hardware Eurorack system. However, setting the **Block DC** to OFF should be used with utter care. A large DC offset directly to your speakers can damage your hearing and/or speakers.



Patch from an **output** *jack, and all* **inputs** *will get highlighted with a* **green** *color.*

While patching from an **input**, *all* **outputs** *will get highlighted with a* **red** *color.*

Understanding Control Voltage

A physical Eurorack modular synth communicates via CV (control voltage), which Modular emulates. CV is used to control parameters such as pitch, cutoff frequency, pulse width and more. An interesting aspect of Eurorack modular synths is that there is no real difference between audio and controller (CV) signals. These can be mixed or used to interact with each other, for example using the output of an oscillator to modulate the cutoff frequency of a filter, so called "Filter FM".

Events in a Modular synthesizer are often triggered by a GATE or TRIGGER pulse. A GATE signal is a pulse that is extended high until released, typically the output of a MIDI TO CV module where the GATE signal corresponds to MIDI note on (GATE rising edge) and MIDI note off (GATE falling edge) messages.

Certain modules requires a short pulse, rather than a GATE. This is called a TRIGGER pulse. Examples of modules that work best with TRIGGER pulses are for example the Heartbeat drum modules (using the MIDI TO TRIGGER module or BEAT SEQUENCER) and SEQUENCER modules.

Interfacing With DAWs And MIDI Controllers

The fact that Modular uses CV also means that it interfaces to DAWs and MIDI controllers in a less integrated—but much more flexible—manner compared to most software instruments. Any connection between Modular and a DAW or MIDI controller requires at least one of the modules in the DAW AND MIDI INTERFACING category in MODULE SELECT VIEW.

In order to use MIDI signals from a DAW or a physical MIDI controller to play notes or control parameters, you need to add one of the MIDI to CV conversion modules to your patch. These modules convert the incoming MIDI data to virtual voltage, which the Modular's modules can react to. You will then need to patch the outputs of these modules to the modules you would like to be affected by the DAW or MIDI controller.

Similarly, to make any of Modular's modules synchronize to the DAW tempo, you need to add the DAW Sync module, and patch that to the sequencer or LFO module you would like to have synchronized. See the detailed description of the DAW AND MIDI INTERFACING modules in the "Modules In Detail" on page 286 and go through the "A Basic Modular Synth Patch" on page 277 to see an example of this in use.

Performance Modules

Both in the studio and during a live performance, a big part of the attraction to using synthesizers is the ability to alter sounds on the fly by adjusting parameters as the music plays. But even with complex patches using a large number of modules, there is usually a limited number of parameters you are interested in tweaking as you go. These parameters can, however, be spread out across a number of different modules that may be placed far from each other in the rack, making live adjustments impractical. To facilitate sound tweaking on the fly, we have included the user definable PERFORMANCE mod-

ules. These have controls (knobs, switches and/or sliders) that you can freely assign to any parameter within your patch. This lets you gather the parameters you would like to play with in a single spot. The parameters become easy to find and easy to assign to a physical MIDI controller.

For more info, please see "Performance Modules" on page 282.

Sequencing

Four different sequencer modules are included in the Modular basic package. You will find them under the SEQUENCER category in MODULE SELECT VIEW. These can be used for analog style step sequencing, and can be synced to your DAW tempo using the DAW SYNC module.

For more info, see "The Sequencer Modules" on page 314.

Modular FX

The Softube Modular can also be used as an audio effect. So instead of basing a patch on an oscillator, you can base it on any audio signal–a guitar, a drum loop or a vocal recording. The only difference is that, as an audio effect, the CENTER ROW will also contain an INPUT stereo pair. This is where the audio enters the Modular FX. Patch and process your incoming audio through any of the Softube Modular effects in order to build your own effects, or just load some of the included Modular FX presets.

Basic Terminology

CV	Control Voltage. A signal that controls other modules.
VCO	Voltage Controlled Oscillator. A module that generates waveforms (makes sound).
VCF	Voltage Controlled Filter. The basic approach to synthesizers is to use a harmonically rich audio signal, such as a sawtooth waveform, and use a filter (VCF) to remove "unnecessary" harmonics. This is called *subtractive synthesis*.
ADSR	Attack/Decay/Sustain/Release. Converts on/off information to a nice envelope with variable attack, decay, sustain and release.
VCA	Voltage Controlled Amplifier. This is a CV controlled volume knob, usually used to apply an ADSR envelope to control the volume of a signal.
LFO/VCLFO	Low Frequency Oscillator, usually Voltage Controlled. Basically the same as a VCO, but for very low frequencies. Use to create vibratos, pulse-width modulations, etc.

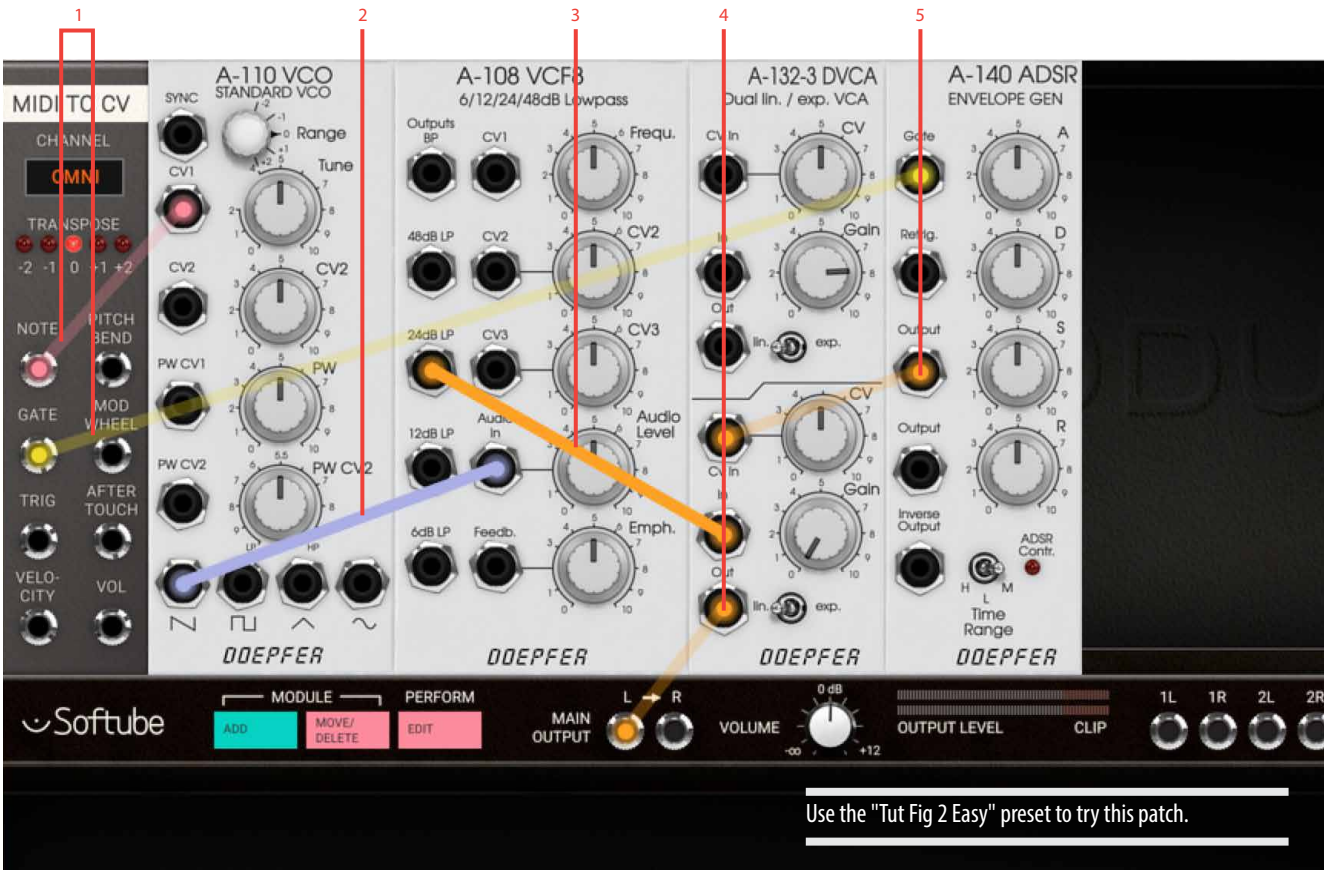
A Basic Modular Synth Patch

This is the smallest patch that uses all of the classic synth parts.

1. The MIDI TO CV sends note information to the tone generating oscillator (VCO), and on/off (**Gate**) information to the envelope generator (ADSR) .
2. The sawtooth output from the VCO is sent to the low pass filter (VCF).
3. The low pass filtered signal is sent to the amplifier (VCA) where its volume will be controlled by the ADSR via the **CV In**.
4. The ADSR converts the on/off information into an envelope with variable **Attack, Decay, Sustain** and **Release**, which is used to control the amplifier (VCA).
5. On the VCA, turn down **Gain** so it is 100% controlled by the CV signal. Finally, the output from the VCA is sent to the **Main Output**.

When sending a MIDI note from your DAW (for example by pressing a key on your MIDI controller keyboard) a note with a slow onset will be heard.

Raise the cutoff frequency (**Frequ.**) on the VCF module fully clockwise to brighten the tone. This can be automated and controlled by the ADSR envelope—connect one of the outputs of the ADSR to the **CV2** jack of the VCF, lower the cutoff frequency again and a familiar filter sweep will be heard each time you strike a note.



Use the "Tut Fig 2 Easy" preset to try this patch.

Basic Sequencer Patch

This is another simple example on how to set up a eight step sequencer to be synced by the DAW to control the simple patch build above. The DAW SYNC module outputs a clock, in this case 16th notes, that is patched into the **Clock In** of the SEQUENCER 8 module.

In order to get the sequencer to start from the beginning when the DAW playback is restarted, the **Reset** pulse jack of the **DAW sync** module is connected to the **Reset In** jack of the **Sequencer 8**. Now, when the DAW is started, the Sequencer will slave to its tempo and output CV, gate and trig information. By connecting the **CV** out to the **CV1** in on the DOEPFER A-110 VCO module, the SEQUENCER 8 steps can control the pitch of the oscillator. When controlling the pitch of an oscillator it is recommended to engage the **Quantize** button on the sequencer module, since this will quantize the outgoing voltage to a chromatic scale. **Gate Out** or **Trig Out** can be used to trigger an envelope generator, such as the A-140 ADSR, used in the example above. The difference between the **Gate Out** and the **Trig Out** is that the **Gate Out's** pulse lengths are longer and that several programmed gates after each other, creates a legato note on the **Gate Out**. In the example above you can see and hear this on step 6 in the sequencer.

Use the "Tut Fig 3 Sequencer" preset to try this patch.

Use the "Tut Fig 4 Beat Sequencer" preset to try this patch.

Beat Sequencer Patch

Using the BEAT SEQUENCER with the DAW SYNC module is more or less the same procedure as the previous example, but with the big difference that this sequencer module doesn't send out any pitch information, only triggers. Another difference is that the BEAT SEQUENCER has four tracks of triggers, each with its own output. These trigger outputs can be used to trigger any kind of event: an envelope (like the A-140 used in the example), another sequencer, a drum module, a clock divider, a sample and hold module et cetera.

In this patch the DOEPFER A-140 ADSR output is used to control not only the volume of the sound (by connecting it to the A-132-2 DVCA's **CV** input), but also to control the A-108 VCF's filter cutoff frequency (by connecting it to **CV2** in), as well as to control the pitch sweep of the A-110 VCO (by connecting it to **CV2** in on the A-110). This creates the synth-tom sound which is triggered from the BEAT SEQUENCER.

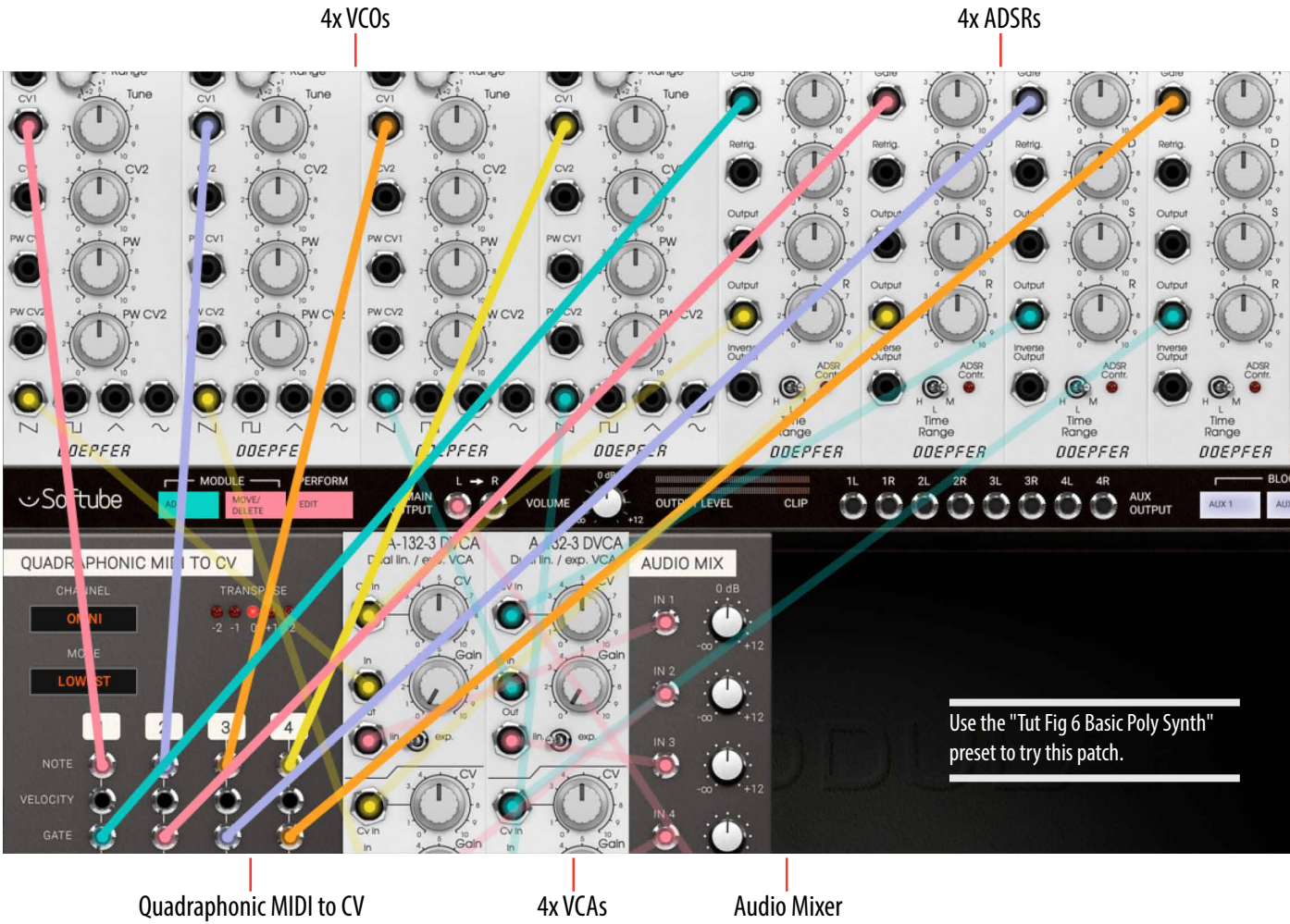
Linking Multiple Sequencers Patch

This example patch uses two linked 16 step sequencers in order to achieve a 32 step sequence. Both 16 step sequencers are clocked from the same clock source (16th notes from DAW SYNC). But they get their reset clocks from the linked out pulse which is transmitted after the last step when the sequencer is not in loop mode. The CV and gates from the two sequencers are mixed together to control the simple synth setup (VCO, VCA, ADSR) at the bottom. The logic tool is used for the logic function OR on the DAW SYNC's **Reset** pulse and the **Link Out** pulse from the second sequencer. This patch has some flaws but it is a good starting point for further patching.

Basic Polyphonic Synthesizer Patch

It's possible to build four voice polyphonic patches with Softube Modular using the QUADRAPHONIC MIDI TO CV module. But just like with a world Eurorack system, this takes four times more modules if each voice needs its own oscillator, filter, etc.

This example demonstrates how a polyphonic patch is set up. Each voice in this really simple patch consists of a VCO going into a VCA that is controlled by an ADSR envelope. Each voice gets its own note CV from the **Quadraphonic MIDI to CV** module and in the same manner, each voice also get its own gate information sent to its dedicated ADSR envelope. The ADSR output is connected to the **CV In** on the VCA controlling the overall volume of that voice. The VCA of each coive is then summed in the **Audio Mixer** before being sent to **Main Output**. Experiment with the QUADRAPHONIC MIDI TO CV voice modes to see the difference between rotating assign and lowest assign modes.



Performance Modules

The PERFORMANCE MODULES offer a nice way of getting easy access to a selection of parameters from all modules in your patch in one convenient location. The modules can be used to create what could be called a macro-area containing all the parameters you want easy accessible for your performance. A Performance knob, switch or slider can be linked to any parameter within your patch, making it possible for you to create a custom interface to your patch.

Performance Linking In Detail

The knobs, switches and sliders on the PERFORMANCE MODULES can be edited to control parameters within your patch, all from one convenient place. Click on the **Perform Edit** button on the CENTER ROW to enter PERFORMANCE EDIT MODE, which is indicated by the activated **Perform Edit** button. Patching is disabled in PERFORMANCE EDIT MODE.

In PERFORMANCE EDIT MODE the knobs on the PERFORMANCE MODULES in your patch that are unassigned will become green tinted, and the text label underneath will also say "CUSTOM VALUE".

By clicking on one the green performance knobs, the knob color turns yellow indicating that you have selected this particular knob for being assigned to a parameter within your patch.

All controls in your patch that are available as destinations, will now be indicated by turning green. Click on the control (knob, switch, slider) within your patch you want as your linked destination. This knob will turn transparent red to indicate that it is linked to a performance knob. The slaved parameter will get disabled as long as it is slaved to a performance module.

1. Click Perform Edit to enter Performance Edit Mode. Available Performance Knobs turn green.

2. Click on a green Performance Knob to select it. Assignable knobs will turn green, and the Performance Knob turns yellow.

3. Click on a green knob to assign it to the Performance Knob.

4. When a knob has been assigned to a Performance Knob it can only be controlled from the Performance Knob, which is indicated by a red color.

5. Hover over the Performance Knob to change to a custom name, or set the ranges of the assignable knob.

The Performance knob name can be edited by hovering your mouse pointer over it until a window appears. In this window, you can write your custom name for the knob's function and also set custom ranges using the **Min** and **Max** knobs.

To unlink a performance knob, double click it and the parameter slaved to it will lose its red color and be available for normal use again.

To exit PERFORMANCE EDIT MODE, click the **Perform Edit** button again and its appearance is reverted back to normal as you're back in PATCH VIEW.

LEDs on Performance modules

LED indicators on Performance modules and its corresponding input jack is for visual reference and can be used for displaying an audio level, gate indication or clock rate. Blinkenlights!

The Preset Library

Modular comes with more than 200 preset patches. Use your DAW's preset browser to browse through the patches. They are organized as follows:

Bass	Different kinds of bass synth sounds.
Brass	Trumpets, horns (OK, we also included some flutes here).
EFX	Effect sounds.
Lead	Different kinds of lead synth sounds.
Pads	Mono and poly chord sounds.
Perc	Different kinds of percussion sounds.
Sequenced	Sequenced and pseudo-sequenced sounds.
Strings	Lush and analog, acoustic and murky.
Tutorial patches	Good starting points, described in the tutorial examples above.
Heartbeat	Sounds that make use of the Heartbeat modules.
Expansion	Sounds that requires add-on modules, for example Intellijel modules.

Artist	Presets created by artists.
Modular FX	Presets made for the insert version of Modular, to be used with an external audio source. These are only available in Modular FX.

Setup Window

Clicking the Setup tab in the bottom right corner of Softube Modular's graphical user interface will bring up the setup configuration menu. Some of the changes made here will only take effect after relaunching your DAW.

Always use smaller GUI	This forces the GUI to remain small even if higher resolution is supported by your computer screen. Uncheck this if you want to use a larger GUI.
Use medium GUI on large screens	This forces the GUI to remain medium sized even if higher resolution is supported by your computer screen. Also uncheck this box if you want to use the largest GUI.
Show value display	Toggles the value display in the lower left corner of Softube Modular on and off.

About CV and Gate Standards

The mid 1960s was an exciting time with huge leaps forward in technology for synthesizing sound. It has since then been widely debated who really invented the voltage control for synthesizers. Nevertheless, voltage control was a significant invention that enabled all synthesizer modules to communicate to each other and it was **Bob Moog** that standardized CV. 1 Volt/Octave for pitch was the first standard to prevail although many other manufacturers for a long time also used other standards such as Hz/Volt (which respond to each octave by a doubling of voltage).

For triggering of electronic switches and envelope generators within his system, Bob Moog used the so called *s-trig* (short trig) standard where others used the *V-trig* (voltage trig) standard. The latter eventually became more popular and is today the standard when we talk about CV (control voltage for pitch) and Gate (for triggering sequences, envelopes etc).

The Eurorack standard, more or less set by **Dieter Doepfer** with his A100 system, use the linear CV and V-trig standard. There are three types of signals within a Eurorack system:

Audio Signals	Ranges from -5 to +5v
Control Voltages	Typically range within 0 to +8v, as for example the output of an ADSR
Trigger Voltages	Range 0 to 5v, trigger occurring at leading edge

All these three signals can be patched without restrictions within the system, which means that audio can be used to trigger logic modules designed for use with trigger voltages or as modulators connected to inputs designed for control voltage. The system will not differentiate between the types which makes the Eurorack system extra intuitive, open for experiments and happy accidents.

Modules In Detail

In this section each module of the Softube Modular is described in detail, with added suggestions on use. In *Description Legend* below we define how each module is described and what the different sections described refers to.

Description Legend

Names in bold
Refers to knobs, switches, dials, input/output jacks, etc.

NAMES IN CAPS
Refers to a parameter setting, or the name of a module or function.

[descriptive text]
Text within brackets is a description of a symbol on the panel, for example that of an waveform or other graphical element.

Parameters

Refers to knobs, dials, switches on module and are described from left to right, from top to bottom on module.

Inputs

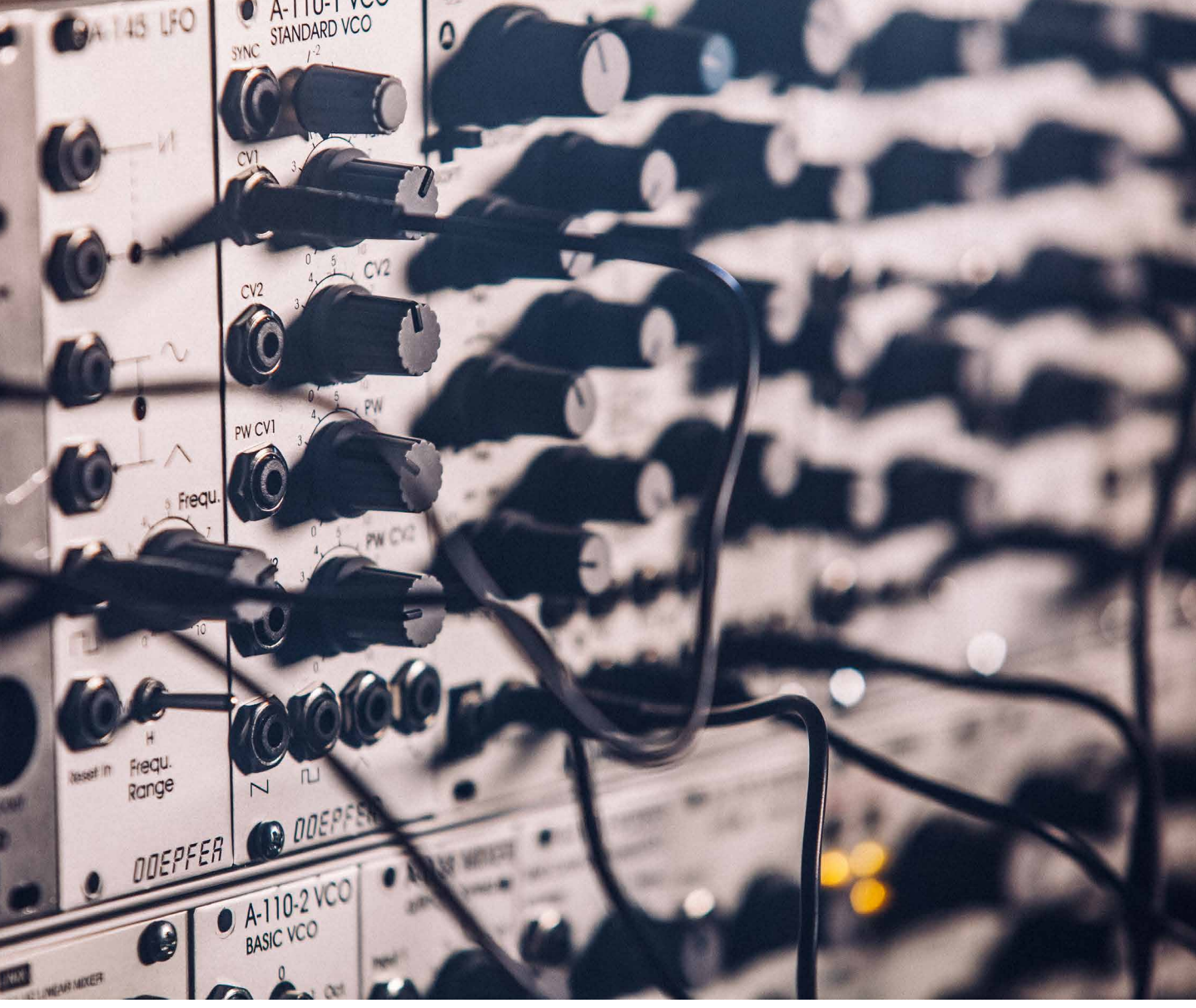
Refers to the input jacks of the module used for input of audio or CV. They are described from left to right, from top to bottom on the module.

Outputs

Refers to the output jack of the module used for outputting audio or CV. They are described from left to right, from top to bottom on the module.

In Use

Detailed description of the module's functions as well as some tips and tricks.

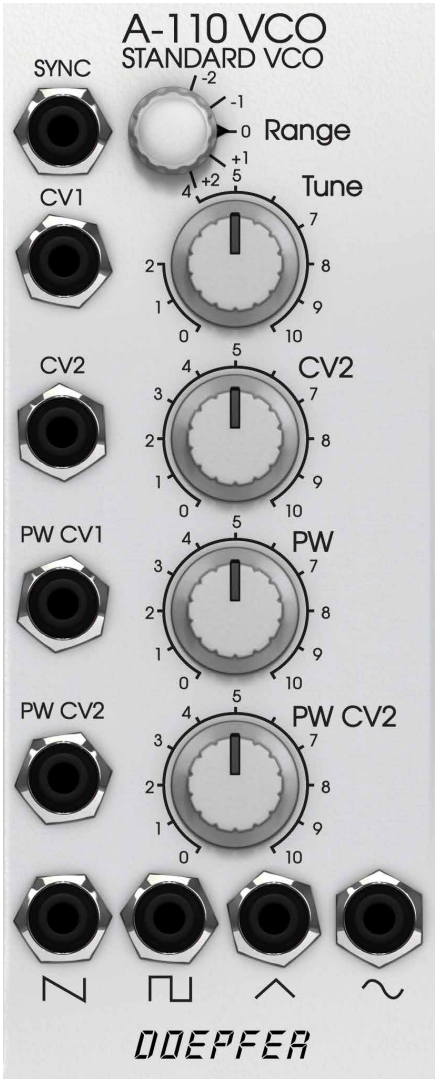


Doepfer

This category contains the modules from legendary Eurorack manufacturer Doepfer. These include a VCO, VCF, VCA, ADSR Envelope, LFO and a Noise source—all you need for building your own basic subtractive synthesizer patch.

Doepfer A-110 VCO (Voltage Controlled Oscillator)

An oscillator is a tone generator, the most basic building block of a synthesizer. The Doepfer A-110 can generate four different waveforms, each with its distinctive own overtone spectrum.



Parameters

Range	Octave offset ranging five octaves
Tune	Fine tune up and down one whole note, preset setting 5 is tuned to 440 Hz when playing an A3 from a MIDI keyboard.
CV2	Sets the amount of CV modulation of the incoming signal at CV2. Knob turned fully clockwise is full 1v/octave range.
PW	This knob sets duty cycle of the pulse wave. The preset setting 5 is 50%, also known as a square wave.
PW CV2	This knob controls the amount of pulse width modulation by incoming CV at PW CV2 jack.

Inputs

SYNC Incoming signal on this jack resets the oscillator, often referred to as hard sync on other synthesizers.

CV1 CV control of pitch, full range 1v/octave. Connect your note CV from MIDI to CV converter module to this jack.

CV2 CV modulation input, connected with CV2 knob (see above).

PW CV1 CV control of pulse width, full range.

PW CV2 CV control of pulse width, connected with PW CV2 knob (see above).

Outputs

[saw] This jack outputs a saw wave, a harsh and clear sound containing both even and odd harmonics.

[pulse] This jack outputs a variable pulse wave, a more hollow sound than the saw wave which contains mostly odd harmonics when the pulse width is 50% (PW knob set to 5).

[triangle] This jack outputs a triangle wave, also a very hollow sound suitable for synthesizing flutes, vibes and organ sounds.

[sine] This jack outputs a sine wave which is a waveform that only contains the fundamental, none of the overtones. This is thus not suitable for subtractive synthesis (shaping the sound with a filter).

In Use

Connect the **Note** CV out from your MIDI TO CV converter to **CV1** on the A110 for accurate full tracking of the A110. By connecting **Note** CV into the **CV2** and using the **CV2** knob, microtonal changes smaller than 1 V/Octave tracking can be obtained. Connect **Sync** jack to the square output of another oscillator to achieve hard sync. This means that A110 is reset by the other oscillator which can be heard when raising the pitch of the A110 above the other one's—the harmonic spectrum of the outputted signal of the A110 is now changed. Control the A110 pitch by envelope, LFO or similar to achieve moving "hard sync sweeps".

The Softube Modular Doepfer A110 module models the real hardware with all its flaws and quirks. This means, for instance, that the triangle and sine outputs are not "mathematically clean" but has the additional harmonics just like the original circuits they are modeled upon. This gives the module character and a nice musicality, but if you prefer a "clean" sine waveform we advise you to use the UTILITY SINE OSCILLATOR instead.

Doepfer A-108 VCF8 (Voltage Controlled Filter)

Voltage Controlled Filter based on the well-known transistor ladder filter (the "Moog" ladder) but with a unique external feedback path.

Parameters

Frequ Cut off frequency of the filter.

CV2 CV2 amount, scales the incoming signal on the **CV2** jack.

CV3 CV3 amount, scales the incoming signal on the **CV3** jack.

Audio Level Incoming volume of the Audio input jack.

Emph Emphasis, this controls the feedback amount of the filter. Often also called resonance or Q.

Inputs

CV1 CV control of cutoff frequency, full range 1v/octave. Connect your note CV from MIDI TO CV converter module to this jack if you want your filter to fully track your keyboard.

CV2 CV control of cutoff frequency. This jack is tied to the **CV2** knob

that scales the amount of the signal from this input.

CV3 CV control of cutoff frequency. This jack is tied to the **CV3** knob that scales the amount of the signal from this input.

You can have several CVs controlling the cutoff frequency, the different CV inputs are added together within the module.

Audio In Insert the signal you want to filter here. This jack is tied to the **Audio Level** knob that sets the volume of the incoming signal.

Feedb This jack breaks up the internal feedback path from the **48 dB** output back via the **Emphasis** knob. This enables different modules to be inserted into the feedback loop. For example, inserting a VCA enables voltage controlled resonance. This also makes it possible to feed back other filter outputs than the 48dB LP to obtain a different resonance behavior.

Outputs

BP This is the bandpass output of the filter, which means that it only passes audio centered around the set cutoff frequency. Emphasis sets the width of the pass band.

48dB LP This is the 48dB per octave low pass filter output, the filter's steepest rolloff.

24dB LP This is the 24dB per octave low pass filter output, often associated with the "fat" American synthesizer sound.

12dB LP This is the 12dB per octave low pass filter output.

6dB LP This is the 6dB per octave low pass filter output, the filter's least steep rolloff which gives it a very gentle impact on filtered sound.

In Use

The Doepfer A-108's audio input is very sensitive, so distortion is possible even with normal levels. For example, distortion appears about from position 5 with the A-110 oscillator. In self-resonating mode, the filter can be used as a sine-oscillator source, tracking 1V/octave on the **CV1** input jack.

Doepfer A-132-3 Dual Linear/Exponential VCA

The A-132-3 is composed of two identical Voltage Controlled Amplifiers where each has a manual gain control and a Control Voltage input with attenuator. The character of the Control Voltage response can be set to either linear or exponential.

Parameters

CV Control Voltage input Amount. This knob scales the incoming CV controlling the VCA.

Gain This knob sets the Gain offset of the VCA. It can be thought of as the "minimum" level or "idle" level setup of the VCA.

Lin/Exp This switch sets the scale response to input Control Voltage at CV in.

Inputs

CV in An incoming signal controls the amplification of the VCA.

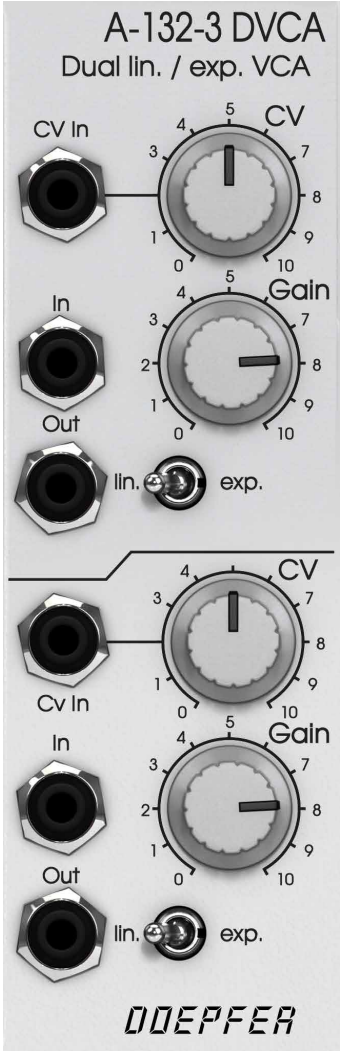
In The input signal which amplitude is to be controlled.

Outputs

Out Output of the VCA, the resulting signal.

In Use

The VCA can be used to shape and scale both audio and control signals. The most common example would be to let an ADSR envelope control the output volume of a synthesized sound in the classic subtractive configuration VCO-VCF-VCA. Other examples include letting the VCA scale the vibrato amount of an LFO to pitch where the CV input is controlled by the Modulation wheel CV from a MIDI to CV module. The VCA can also be used to create tremolo effects when controlled from a LFO or used with the A-108 VCF8 to enable it to have Voltage Controlled feedback.



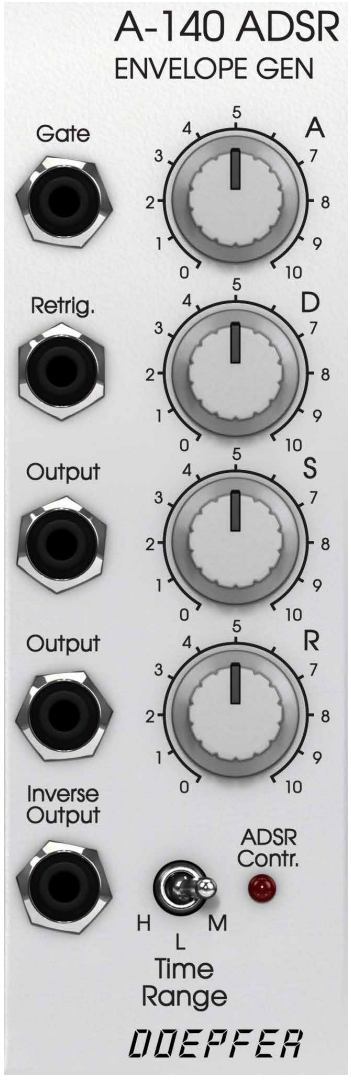
Doepfer A140 ADSR Envelope Generator

An ADSR envelope generator is a gate-triggered time-controlled series of events outputting timed changes in voltage output. The different phases represents that of an acoustic sound described in its **Attack**, **Decay**, **Sustain** and **Release** phases.

The **Attack** phase is the swelling phase, from zero to the maximum value of the envelope. The **Decay** phase is the subsequent falling time period until the **Sustain** level is met. The **Sustain** level determines the level as long as the MIDI key (or gate voltage) is held. The **Release** phase is the ending time period occurring when the key (or gate voltage) is released and determines the time it takes until the envelop have reached zero again; the end point.

Parameters

A (Attack)	This knob sets the *Attack* time of the envelope.
D (Decay)	This knob sets the *Decay* time of the envelope.
S (Sustain)	This knob sets the SUSTAIN level of the envelope.
R (Release)	This knob sets the RELEASE time of the envelope.
H L M (Time Range)	This 3-position switch lets you select the overall speed of the ADSR. H: HIGH, very slow. Up to several minutes.



M: MEDIUM, standard setting.

L: LOW, very short time periods down to less than 100 µsec.

Inputs

Gate	This input triggers the ADSR envelope from a gate signal.
Retrig	This input retriggers the envelope from a trigger signal.

Outputs

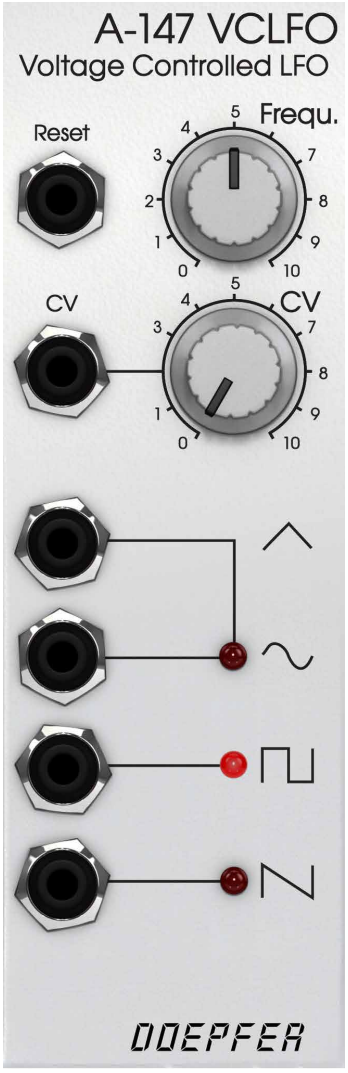
Output	This is the output voltage from the envelope.
Output	This is a duplicate of the output above.
Inverse Output	This is an inverted version of the **Output**.

In Use

The Doepfer A-140 ADSR is a module that is triggered from a gate generated from for example a key pressing (using the MIDI TO CV module) or a sequencer. **Attack** and **Decay** sets the time of the first two stages before arriving at the set **Sustain** level. **Release** is the fall time from **Sustain** level when gate has been released. The ADSR envelope can be used for controlling the VCA, VCF or VCO. Also use the ADSR is to sweep the pitch of a VCO synced to another VCO for sync-sweeps.

Doepfer A-147 Voltage Controlled LFO

A LFO is an oscillator especially adapted to be used as a modulation source running at a slow rate. An LFO is often used to create trills, vibrato, tremolo or control of slowly evolving soundscapes.



Parameters

Frequ This knob sets the initial speed of the LFO and ranges from 0.01 Hz to 50 Hz.

CV This knob sets the Control Voltage amount that the signal on the CV input will affect the LFO speed.

Inputs

Reset This input jack resets the LFO on a rising edge signal such as a Gate signal.

CV Input of Control Voltage affecting LFO speed.

Outputs

[triangle] Triangle wave output, suitable for vibrato use (modulating pitch).

[sine] Sine wave output, also suitable for vibrato and tremolo.

[square] Square wave output, suitable for trills (modulating pitch).

[sawtooth] Sawtooth wave output, suitable for creating a beating kind of motion in your sound.

In Use

The most obvious use for the Doepfer A-147 is of course as a modulation source for creating vibrato (modulating pitch) or tremolo (modulating amplitude). By connecting the A-147 **Sine** or **Triangle** output to a CV in of an oscillator a vibrato can be obtained.

Connect the mod wheel or after touch output of the MIDI TO CV module to the **CV** input to control the amount from your MIDI keyboard.

Using the A-147 to slowly sweep the cutoff frequency of your filter is another great way of creating timeless and classic synth sounds.

Another fun thing to do is to use multiple A-147 LFO resetting each other to create complex and "pseudo-sequenced patterns".

Doepfer A-118 Noise + Random Voltage

Module A-118 is a noise and random voltage generator. It produces three types of signal: white noise, colored noise, and random voltage.White and colored noise can be used as audio sources, and the random voltage is a useful source of voltage control, especially for its low frequency content.

Parameters

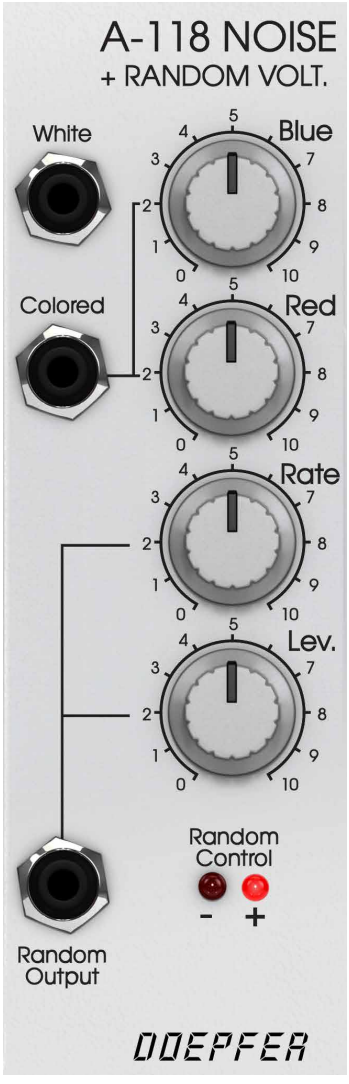
Blue	This knob boosts the high end of the colored noise output.
Red	This knob boosts the low end of the colored noise output.
Rate	This knob controls the slew rate of the random output. Lower values gives less slew and thus more vivid changes in the random output.
Lev	This knob controls the output level of the random output.

Outputs

White	This is the white noise output.
Colored	This is the colored (equalized) noise output.
Random Output	This is the random noise output, it consists merely of low end heavily filtered noise.

In Use

The Doepfer A-118 is a dirty little noise module emulating the amplified noise of a transistor with additional outputs for colored noise for that eerie wind sound. Use **Random Output** to breathe life into boring patches and use **Blue** noise in combination with a clocked sample and hold module for that classic droid sounds.



Doepfer A-114 Ring Modulator

The Doepfer A-114 Ring Modulator contains two separate ring modulators. Each ring modulator outputs the product (Multiplication X*Y) of the signals at inputs X and Y. It behaves similar to a VCA but responds to both positive and negative voltages (4-quadrant multiplication). Ring modulator could therefore be described as a version of amplitude modulation (AM). Where ordinary amplitude modulation outputs the original carrier frequency fC as well as the two side bands (fC - fM, fC + fM), ring modulation cancels out the carrier frequencies, and just lets the side-bands pass to the output.

The emulation of the A-114 Ring Modulator featured in Softube Modular reflects the original hardware slight variation of trimming between the top and bottom channels. The top channel does not fully suppress the X component out of the X*Y output as the bottom channel are. The distortion models also differ slightly between the two to mirror the behavior of the hardware measured.

Inputs

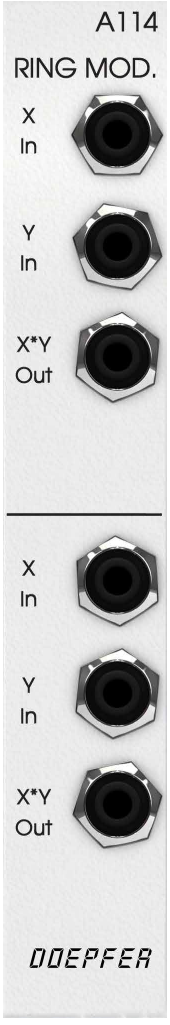
X in 1	input for audio signal X, top channel.
Y in 1	input for audio signal Y, top channel.
X in 2	input for audio signal X, bottom channel.
Y in 2	input for audio signal Y, bottom channel.

Outputs

X*Y out 1	audio output for the top channel.
X*Y out 2	audio output for the bottom channel.

In Use

A ring modulator is great for the production of bell-like sounds, alien voices, or just to produce new timbres out of exciting components.

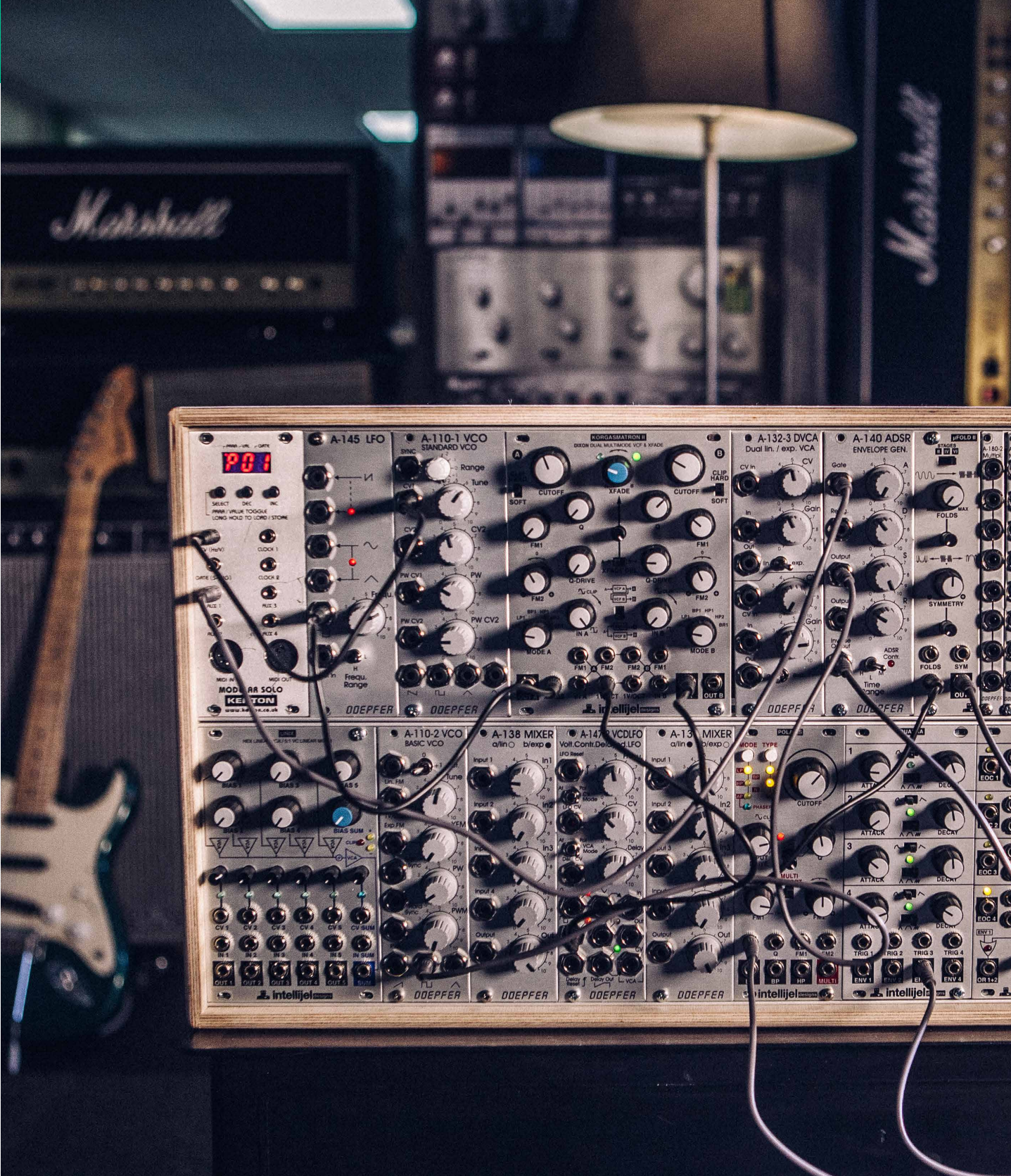


Intellijel

Eurorack brand Intellijel, run by **Danjel van Tijn** and associates, is based in Vancouver, British Columbia in Canada. These branded add-on modules from Intellijel are available for purchase at `http://softube.com/buy`.

The Intellijel modules are RUBICON THROUGH ZERO OSCILLATOR, KORGASMATRON II and the µFOLD II. These three modules all make a fine addition to the more traditional subtractive basic system of Softube Modular, as well as a step towards more West coast thinking in terms of synthesis.

For more information regarding the Intellijel modules, please see `"27 Intellijel Rubicon, Korgasmatron II & uFold II" on page 236`.


A-145 LFO
A-110-1 VCO
STANDARD VCO
KORGASMATRON II
A-132-3 DVCA
Dual lin. / exp. VCA
A-140 ADSR
ENVELOPE GEN.
A-110-2 VCO
BASIC VCO
A-138 MIXER
intellijel
DOEPFER

ROLI Seaboard RISE Module

Interface the innovative keyboard controller **ROLI Seaboard RISE** with Modular and get access to five dimensions of touch.

Configuration

Please make sure that ROLI Seaboard RISE is set to its default settings. To load these, start the application ROLI DASHBOARD FOR RISE. Press the MENU button on the top right, and then select RESET in popup menu that appears.

The RISE's **Power/Mode** button on the lower left side should now be WHITE, to indicate the RISE is in MIDI mode. If your Seaboard is not in MIDI mode (the button has a different color), press the button repeatedly until it turns white.

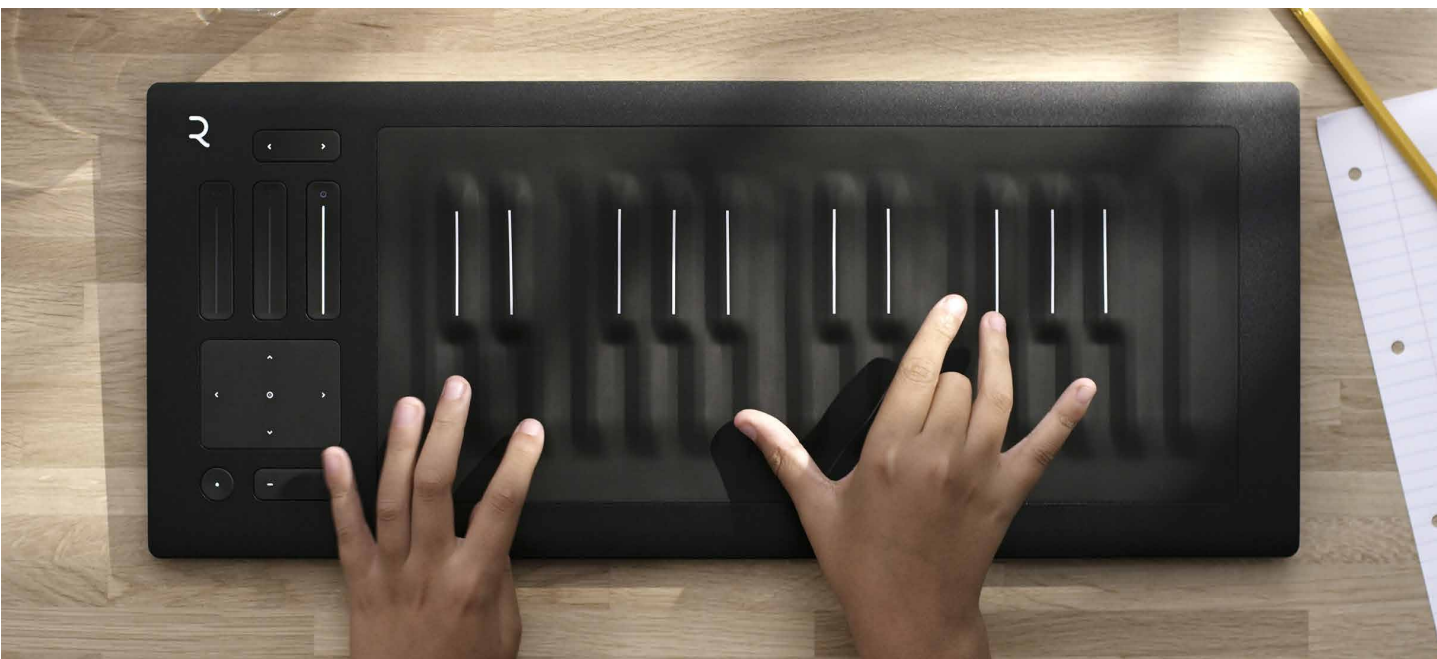
Set up your DAW for working with the RISE as described in detail on ROLI's web site: `https://support.roli.com/article-category/rise-setting-up-with-other-software`

This is particularly important to obtain the correct behavior in Ableton Live as described here:

`https://support.roli.com/article/using-the-seaboard-with-ableton-live`

Parameters

Glide to Note Turns on and off the RISE's ability to glide between notes. Turning this parameter OFF does not only disable the gliding zones beneath and above the RISE keybed, but also disables the vibrato (**Glide** expression) achieved by striking a note and then wiggling your finger sideways on the playing surface. Turning this parameter OFF is advisable for situations where you want the RISE to hit precise chromatic pitch values and don't want to glide between notes (or want to use the glide to control any other parameter).



Mode POLY ROTATE: Cyclic voice allocation, a new voice is assigned with each received MIDI note from RISE.

POLY LOWEST: Only one voice channel is triggered as long as only one note is played at the time. If one voice is playing (key is being held, gate is high), the next note added will assign voice channel two and so on.

MONO: In this mode, only channel 1 is triggered by playing RISE.

Indicators

Voice LED 1-4, All Indicates which voice is currently being triggered by your RISE. All is lit if any voice is active.

Touch Faders 1-3 Indicates the value set on the Touch Faders 1-3. These faders are active in RISE's MIDI mode (the **Power/Mode** button on the lower left on RISE is white) and sends out CV ranging from 0 to 5V via the jacks CV1-3 (see more on the CV 1-3 outputs below).

XY Touchpad Indicates the value set by the XY Touchpad on your RISE. These are connected to the X and Y outputs which send out a corresponding control voltage, ranging from 0 to 5V (see more on the X and Y outputs below).

Outputs

CV 1-3 These jacks send out CV, 0 to 5V, controlled by the **Touch Faders 1-3** on your RISE.

Pedal This jack sends out CV, 0 to 5V, controlled by an expression or sustain pedal connected to the RISE pedal jack.

X and Y These jacks sends out CV, 0 to 5V, controlled via the **XY Touchpad** on your RISE.

Voice Outputs 1-4 These are the polyphonic outputs for CV and Gate information. Voice assign set by the **Mode** parameter will affect the way these output jacks are assigned.

Note This is the note CV output (1V per octave) for each of the four voices. Connect this output to the main tracking input on your oscillator (such as CV1 on DOEPFER A-110). When the **Glide to Note** function is activated, this output is not only affected by the played MIDI key but also the RISE channel's glide.

Gate These jacks output a logical high signal (5V CV) when receiving a MIDI note on message (a key is struck on your RISE). At MIDI note off (key is released) it instantly falls back to zero (0V CV). Use this output to connect to a **Gate** input on an ADSR envelope module.

Strike This jack outputs 0 to 5V CV reflecting the velocity value (0–127) received from the RISE. The harder you strike the RISE keys, the higher CV output you will get here.

Press This jack outputs 0 to 5V CV when receiving pressure from the RISE. This means that Modular can receive polyphonic pressure via the RISE module.

Glide The **Glide** jack outputs positive or negative CV (1V per octave), corresponding to the horizontal upwards or downwards bend from the note played. The **Glide** jack is useful as a modulation source especially when the **Glide to Note** function is set to OFF.

Slide The **Slide** jack outputs CV matching the vertical slide movement performed on your RISE keys.

Lift This jack outputs CV matching the speed of release on a released key on the RISE. The faster you release a played note, the higher the CV output here will be.

All Gate The **All Gate** jack output sums the individual channel gate outputs to a single mono gate output. A

logical high signal (5V CV) is sent when receiving a MIDI note on message (a key is struck on your RISE).

All Press This jack outputs the average pressure CV from all the channels. This means that the more fingers used and the more pressure applied on the RISE's playing surface, the higher the CV output will be.

All Glide The **All Glide** jack outputs positive or negative CV (1V per octave), corresponding to the average horizontal upwards or downwards bend from the notes played.

All Slide The **All Slide** jack outputs CV corresponding to the average vertical slide movement performed on your RISE keys. One useful usage of the **All Slide** output is to control the filter cutoff in a paraphonic patch (a patch with multiple oscillators but only one filter). This makes it possible to control filter sweeps by moving all of your fingers vertically on the RISE.

In Use

When interfacing with the Seaboard RISE in Modular through Modular's RISE module, it's good practice to use SLEW LIMITER modules to soften the sudden shifts in CV out, which can otherwise result in audible clicks and undesirable behavior. We recommend that you study the RISE preset patches included among Modular's presets for inspiration.

The expressiveness of the RISE makes it possible to make really organic sounding patches using for example pressure, strike, glide, slide and lift. The **XY Touchpad** is really useful when controlling panning and volume at the same time. The **Touch Faders 1-3** are useful for example when creating drawbars on an organ or controlling volumes on different channels.

DAW and MIDI Interfacing

These modules are mainly for interfacing the Softube Modular with your DAW. It features DAW SYNC (MIDI to clock converter), MIDI TO CV (monophonic MIDI to note and gate converter), QUADRAPHONIC MIDI TO CV (four channel, polyphonic MIDI to note and gate converter) and MIDI TO TRIGGER (four MIDI notes to trigger outputs, for triggering of percussive sounds from individual MIDI notes).

DAW Sync

This module is an efficient way of getting clock pulses in sync with your DAW. Connect the 16 notes output to click in on your sequencer module and the reset pulse out to reset in for the most common application of this module.

Outputs

1/1	This jack outputs a short pulse on the first beat of each measure.
1/4	This jack outputs a short pulse on each beat (quarter notes).
1/8	This jack outputs a short pulse on each eighth note.
1/16	This jack outputs a short pulse on each sixteenth note.
1/32	This jack outputs a short pulse on each thirty second note.
Reset	Sends out a trigger pulse at the beginning of a bar in the DAW's sequencer.
Run	This jack sends out a high (+5v) signal, when DAW is running.

In Use

Typical use for the DAW SYNC module is to supply clocks for the sequencer modules. Connect for instance 1/16 output to clock in on the sequencer module and the **Reset** out jack on DAW SYNC to **Reset** in on the sequencer.

The **Run** jack can be used together the LOGIC TOOL module with for example a gate from a sequencer before sending to an envelope GATE IN. In this way, the ADSR gate input is sure not to be held high when DAW is stopped.

MIDI to CV

Your average workhorse MIDI to CV converter! This module converts incoming MIDI notes to monophonic **CV** and **Gate** outputs. There is also a trigger out for resetting oscillators with each keystroke if that's to your liking. Outputs for the most common performance MIDI controllers such as pitchbend, mod wheel, volume and aftertouch are also provided.

Parameters

MIDI channel This sets the MIDI to CV receiving channel. OMNI is for receiving on all channels, while numbers 1 to 16 reflects specific MIDI channels.

Transpose Transpose incoming MIDI notes, plus and minus one or two octaves.

Note Priority Set the note pitch priority when playing several MIDI notes at once. LOW PRIORITY means lower MIDI notes are selected over higher, while HIGH PRIORITY has the opposite behavior. LAST note priority simply means that the last received MIDI note is selected for output note pitch. Note that this only applies for note CV, not velocity CV, since velocity always has the last note priority.

Outputs

Note 1 volt per octave note CV output. Use this output to connect to your main tracking input on your oscillator (**CV1** on A-110 for example).

Pitch Bend This jack outputs +/- 1V CV (+/- 1 octave) when receiving MIDI pitch bend.

Gate This jack outputs a logical high signal (5V CV) when receiving a MIDI note on message (a key is stroke on your keyboard) from DAW. At MIDI note off (key is released) it instantly falls back to zero (0V CV).

Use this output to connect to a **Gate** input on an ADSR envelope module. When receiving several MIDI note on messages, the gate will remain high (at output 5V CV) until last note is released with a MIDI note off message (your key on your keyboard is released).

Mod Wheel This jack outputs plus 1V CV when receiving MIDI modulation (CC #01).

Trig This jack outputs a short logical high pulse (5V CV) when receiving a new MIDI note on message from the DAW.

Aftertouch This jack outputs 0 to 5V CV when receiving a MIDI aftertouch message (channel pressure) from DAW.

Velocity This jack outputs 0 to 5V CV reflecting MIDI velocity value (0–127). The harder you play on your keyboard, the higher CV output you will get here.

Vol This jack outputs plus 0–5V CV when receiving MIDI volume (MIDI CC#07).

In Use

Note and **Gate** outputs can be used to hook up your basic monophonic synthesizer patch, as described in the first tutorial example (see above). **Trig Out** can be used to reset oscillators and LFOs, or retriggering of ADSR envelopes such as the A-140. Velocity and aftertouch outputs can be used to control filter cutoff modulation or amplitude.

Quadraphonic MIDI to CV

This is a four channel MIDI note to CV and gate converter for creating those four voice polyphonic patches that would cost a fortune in real life.

Parameters

Channel This sets the MIDI to CV receiving channel. OMNI is for receiving on all channels, while numbers 1 to 16 reflect specific MIDI channels.

Transpose Transpose incoming MIDI notes, plus and minus one or two octaves.

Mode This sets voice allocation mode. ROTATE is cyclic voice allocation, so that a new voice is assigned with every MIDI note received from DAW. LOWEST is voice allocation that assigns only voice channel one as long as only one note is played at a time. If one voice is playing (key is still being held, gate is high), next note added will assign voice channel two and so on.

Outputs

Note 1 volt per octave note CV output. Use this output to connect to your main tracking input on your oscillator (**CV1** on A-110 for example).

Velocity This jack outputs 0 to 5V CV reflecting MIDI velocity value (0–127). The harder you play on your keyboard, the higher CV output you will get here.

Gate These jacks output a logical high signal (5V CV) when receiving a MIDI note on message (a key is stroke on your keyboard) from the DAW. At MIDI note off (key is released) it instantly falls back to zero (0V CV). Use this output to connect to a **Gate** input on an ADSR envelope module.

Volume This jack outputs plus 1V CV when receiving MIDI volume (MIDI CC #07).

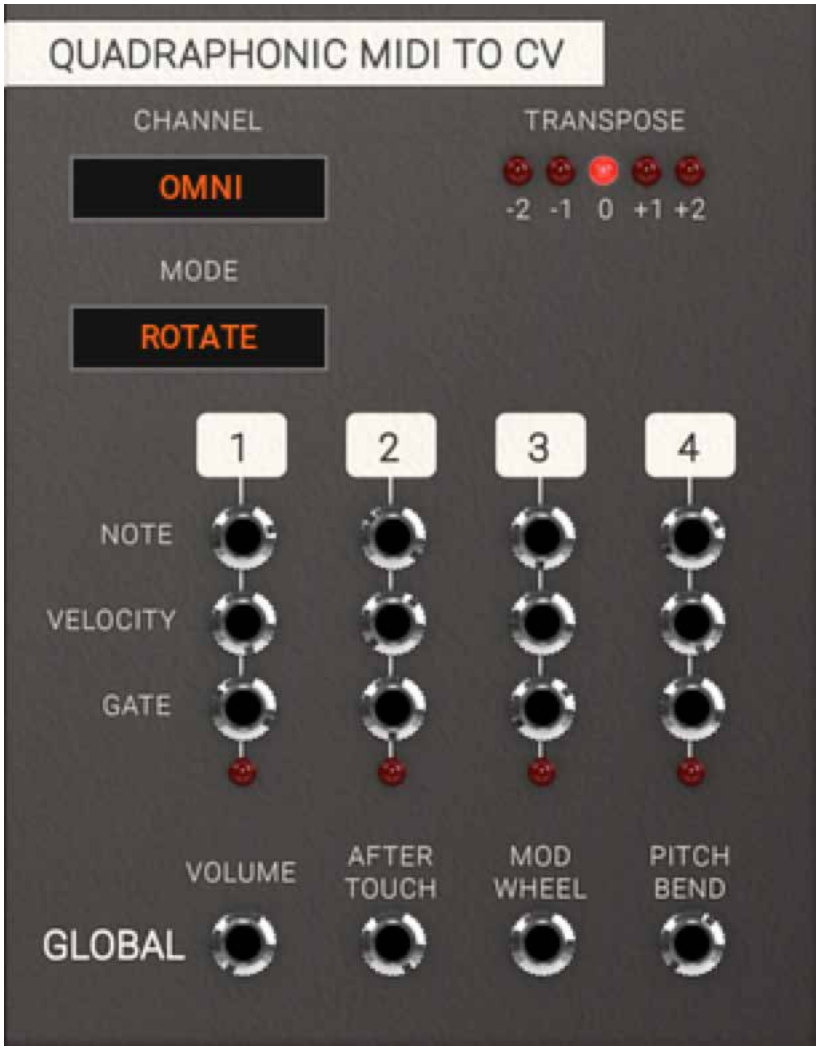
Aftertouch This jack outputs 0 to 5V CV when receiving MIDI aftertouch message (channel pressure) from DAW.

Mod Wheel This jack outputs plus 1V CV when receiving MIDI modulation (CC #01).

Pitch Bend This jack outputs +/- 1V CV (+/- 1 octave) when receiving MIDI pitch bend.

In Use

Connect the **Gate** outputs to four ADSR modules controlling VCAs and the **CV** outputs to four VCOs and you have the basics for a four voice synthesizer (see tutorial example "Basic Polyphonic Synthesizer Patch" on page 281. The QUADRAPHONIC MIDI TO CV can also be used together with the signal tools to create a simple four tone arpeggiator (see for example the preset *Seq Melodic Arp Mono*).



MIDI to Trigger

A four channel MIDI to trigger module, primarily made to trigger percussive sounds off specific MIDI notes. It emits short trigger pulses when a MIDI note is received on the set MIDI channel.

Parameters

Channel This sets the MIDI to CV receiving channel. OMNI is for receiving on all channels, while numbers 1 to 16 reflect specific MIDI channels.

Note This is the specific MIDI note which the MIDI TO TRIGGER will receive on this trigger channel. The **Activity LEDs** will blink when a MIDI note message is received on the corresponding channel.

Outputs

Trigger This jack outputs a short logical high pulse (5V CV) when receiving a MIDI note on matching the one set on this channel.

Velocity This jack outputs 0 to 5V CV reflecting MIDI velocity value (0–127). The harder you play on your keyboard, the higher CV output you will get here.

In Use

Set desired receiving MIDI channel and MIDI notes by clicking and dragging. **Trigger Out** emits short trigger pulses and **Velocity** outputs last known velocity CV, which is held until next MIDI note is received on the channel.

Effect

This category currently only contains the DELAY module, but will be definitely be expanded upon in the future.

Delay

This module delays an incoming signal up to 1000 ms (one second).

Parameters

Time	This knob sets the time between 1 ms to 1000 ms (one second).
CV Amount	This knob sets the CV Amount affecting the delayed time of the inputted signal.

Inputs

In	Use this input for the audio or CV you want delayed.
CV	This is input is used for control voltage of delayed time. Closely linked to CV Amount.

Outputs

Out	Output of delayed signal.

In Use

Use the DELAY module to delay any audio or CV signal. Connect it to a mixer and feedback it through a filter to create a classic tape style effect.

Short delays swept in time from a LFO can also be used to create flangers, chorus and such effects. Simple but powerful module.

Mixer Modules

These are mixers suitable for summing audio (exponential response) and CV (linear response) as well as a polarizing mixer suitable for subtractive mixing, but of course all mixers can be used for either type of signal. All mixer modules have the same controls.

Audio Mixer

Mixer module with an exponential response, making it suitable for mixing audio signals. This mixer uses a dB scale for its knobs, with a maximum gain of 12 dB.

CV Mixer

A no-nonsense CV mixer with a linear response, making it suitable for mixing CV signals. This mixer scales the signal up to 4 times.

Polarizing Mixer

A mixer module with a gain -1 to +1.

Parameters

Volume 1-4	These volume knobs set the amplification of input signal. from -∞ to 12 dB, from 0 to 4x, and from -1 to +1 in the different mixers.
Output Volume	Sets the main output mix amplification. Same range as the input **Volume** knobs.

Inputs

Input Jacks 1–4	The four input jacks for the signals to be summed

Outputs

Output	Mixed and amplified output of the four input signals.

Audio Mixer In Use

The AUDIO MIXER is of course preferably used for mixing audio signals, however nothing prevents you from using it for mixing of any kind of signals since there is no DC block on the input or output.

CV Mixer In Use

The CV MIXER is of course preferably used for mixing CV signals, however nothing prevents you from using it for mixing of any kind of signals.

Polarizing Mixer In Use

Use the POLARIZING MIXER for adding and subtracting signals. The POLARIZING MIXER is great to for example mix the outputs of the A-108 to create new and interesting filter characteristics. It is also useful for mixing control voltages or just about any kind of signals.

The Sequencer Modules

These are simple 16, 8 and 5 step note sequencers, as well as the BEAT SEQUENCER, a four channel trigger sequencer suitable for triggering percussive elements. All are driven by clock inputs that can be taken from the DAW SYNC module or whatever kind of pulse output.

Beat Sequencer

Classic x0x style sequencer for programming percussive sounds (for example with the HEARTBEAT modules, purchased separately). The sequencer can be looped or not. In LOOP OFF mode the **Link Out** sends a trig after last step which can be used to link together several sequencers. SWING sets the delay time of every other pulse input.

Parameters

Gate/Trig Buttons 1–16 — These buttons determine whether or not a trig is outputted on the corresponding output jack.

Indicator LEDs 1–16 — These LEDs indicate which steps are active.

Length — This parameter sets the final step played in the sequence. Change the value by clicking on it to add one step at the time, or click and drag to set whatever value between 2 and 16 you want.

Loop — This button determines whether the sequence will loop back after the last step or not. An unlit **Loop** button means no loop back, and that the **Link Out** jack sends a pulse after the last step has been reached and one more click has been received at the **Clock In** jack.

Swing — This knob sets the delay time of each other pulse input. **Swing** is also dependent on reset pulse in, which can be used for interesting and surprising effects.

Inputs

Clock In — By sending a short pulse or gate into this input jack, the sequencer will progress one step. The most common application of this jack is to connect it to **1/16 Output** of the DAW SYNC module.

Reset In — By sending a short pulse or gate into this input jack, the sequencer will instantly reset to the first step. The most common use is to connect it to the **Reset Out** or **1/1 Output** of the DAW SYNC module.

Outputs

Trig Out — This output jack sends short pulses for each activated step. This type of triggers is useful for triggering percussive elements, using for example the HEARTBEAT modules.

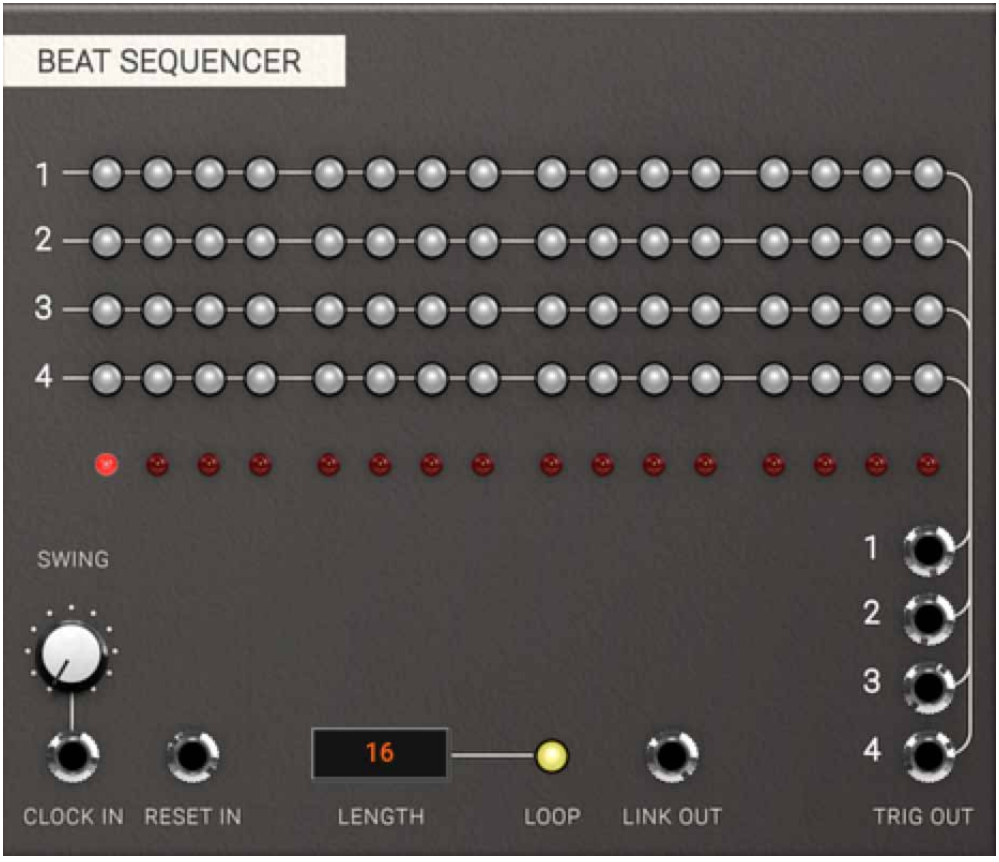
Link Out — This jack outputs a short pulse after the last step has been reached when the module isn't in LOOPED

mode and the sequencer receives another clock in. Use it to reset the next sequencer you want to link in order to expand your sequence beyond 16 steps.

In Use

Patch **Trig Outs** to **Trig Ins** on HEARTBEAT modules or to **Gate In** on the ADSR envelope you want to trigger.

To create accents, use one channel to control an envelope module and a VCA and run your sounds through that to control their dynamics. **Swing** means that every second clock pulse gets delayed, so if you for example use the 16th notes output from DAW sync to feed the clock input on the sequencer, only the steps with equal numbers (2,4, 6 etc) will be affected. This becomes apparent when programming pulses on all 16 steps.



Sequencer 16, Sequencer 8 and Penta Sequenza

Classic step sequencers with **CV Out** and **Gate Out**. Multiple gates after each other extends the gate output over several steps. A separate trig channel is provided as well as chromatically quantized CV scaling. Sequences can be looped or not.

Parameters

Indicator LEDs 1–16 These LEDs indicate which steps are active..

CV Knobs 1–16 These knobs set the amount of output CV at the **CV Out** jack when the corresponding step is active. CV AMOUNT is shown in volts down in the value display in the lower left side of the plug-in.

Gate/Trig Buttons 1–16 These buttons determine whether a gate and a trig is outputted on the **Gate Out** and **Trig Out** jacks when the corresponding step is active. Several consecutive gates creates legato notes, but separate triggers.

Length This parameter sets the length of the sequence. Change the value by clicking on it to add one step at a time, or click and drag to set the value between 2 and 16.

Loop This button determines whether the sequence will loop after last step or not. If **Loop** is OFF, **Link Out** will send a pulse after a completed sequence.

Range This button switches the **CV Out** between limited range (1 octave) and full range (5 octaves). When activated, full clockwise position of the CV knobs will correspond to a one octave jump.

Quantize When activated, the **CV Out** jack will be quantized to the nearest semi tone. Use this function when using the sequencer to program note information.

Inputs

Clock In By sending a short pulse or gate into **Clock In**, the sequencer will progress one step. The most common application is to connect the **1/16** output of the DAW SYNC module to the **Clock In**.

Reset In Reset the sequencer step by sending a pulse to this jack. Most common application is to connect the **Reset Out** or **1/1** output of the DAW SYNC module to the **Reset In**.

Outputs

Trig Out Sends short pulses for each activated step that the sequencer passes when clocked. These kind of triggers are useful for triggering percussive elements, such as the HEARTBEAT modules.

CV Out Sends the programmed CV for each step. **Range** button and **Quantize** button affects the output CV information (see above).

Gate Out Sends a gate for each activated step that the sequencer passes when clocked. Several consecutive gates creates a combined long gate (legato).

Link Out This jack outputs a short pulse after the last step has been reached when the module isn't in LOOPED mode and the sequencer receives another clock in. Use it to reset the next sequencer you want to link in order to expand your sequence beyond 16 steps.

In Use

When linking more than one sequencer, use LOOP OFF mode, where the **Link Out** sends a trig after last step, which can be used to link together several sequencers. Use the **Link Out** trigger to reset the next sequencer, also in LOOP OFF mode, clocked from same clock source (see "Linking Multiple Sequencers Patch" on page 280). The sequencer can be used not only for note information, but also as a programmable modulation source.

The PENTA SEQUENZA is ideal for use as a quirky modulation sequencer. Using an odd number of stages for running a more traditional sequencer like the 16 step sequencer in parallel can create some really interesting polyrhythmic effects.

The Utility Modules

The Utility Modules is a collection of simple useful modules, such as a simple SINE OSCILLATOR, an ENVELOPE FOLLOWER, MULTIPLE, OFFSET, SLEW LIMITER, CLOCK DIVIDER, LOGIC and SIGNAL TOOL, as well as SAMPLE & HOLD and CV-controllable switches.

Sine Oscillator

This is a simple and mathematically correct sine oscillator with perfect tracking. It was originally created for testing purposes in the early stages of the development of Softube Modular.

Parameters

Pitch This knobs sets pitch offset ranging from 0.1 Hz up to 10 kHz. The 12 o'clock position is a C at 64 Hz (this position is easily reached by ALT clicking on the pitch knob).

Inputs

CV in This is a normal 1 V/octave tracking input jack. Connect note CV out from the MIDI TO CV module for musical use of the sine oscillator.

Sync This is a reset input jack for syncing purposes. Similar to hard sync on other oscillators.

Outputs

Out This is the main output jack.

In Use

The sine oscillator is great as a space efficient LFO or as FM operator in use with for example the INTELLIJEL RUBICON.

Envelope Follower

This module can be used to generate control voltages from audio sources, but also as a simple ASR envelope.

Parameters

Attack This knob controls how quickly the envelope follower will react to a rising edge amplitude change.

Release This knob controls how fast the envelope follower will react to a falling edge amplitude change.

Inputs

In This is the input jack. Connect it to audio or gate signal from your MIDI to CV module if you're using it as a regular envelope.

Outputs

Out This is the output jack. Resulting CV from the envelope follower is output here.

In Use

Use the envelope follower to, for example, create an auto-wah by letting audio controlling a low pass filter. The envelope follower can also be used as a simple ASR (Attack, Sustain, Release) envelope if it's triggered with a gate signal.

Multiple

This module multiples the incoming signal to four identical outputs.

Inputs

In	The input signal.

Outputs

Out 1 - 4	Four identical outputs for distribution of the input signal to four different sources.

In Use

The MULTIPLE can be seen as quite unnecessary in the Softube Modular since all outputs can be routed to any number of inputs, but it can be used for esthetic reasons, just to keep your patch tidy. It is also true to how many real life Eurorack modular systems look like when not using "stackables" (stackable cables).

Offset

The OFFSET module can be used both as a static CV source, but of course also to offset the input CV.

Parameters

Offset This knobs sets offset between -10V through 0V to a maximum of 10V.

Inputs

In Input signal to be offset goes here.

Outputs

Out This is the sum of the input signal and the set offset. If a signal is not inserted at input, output will directly correspond to set offset.

In Use

Offsets can be used to set level threshold, for instance with the A>B function in signal tool. Offsets can also be used to control several CV controllable parameters at the same time and linked to a performance knob for "morph-type" operations.

Quad Offset

More or less the same as OFFSET, but with four channels contained within the same module to save space.

Parameters

Offset 1- 4 These knobs set the offsets of the four channels between -10V through 0V to a maximum of 10V.

Inputs

In 1 - 4 Input signals to be offset goes here.

Outputs

Out 1- 4 These jacks output the sums of the input signals and the set offsets. If signals are not inserted at input, the outputs will directly correspond to set offset.

In Use

More or less the same areas of use as the single offset module.

Sample and Hold

The SAMPLE AND HOLD module listens to an incoming signal and samples the control voltage at a time set by the incoming trig pulse at the trigger input.

Inputs

In	Input signal to be sampled from. Can be any signal imaginable.
Trigger In	Pulse input. Input signal at the **In** jack is sampled at rising edge of a pulse.

Outputs

Out	Outputs the sampled and held CV signal. CV is held until next trigger pulse is presented at the trigger input.

In Use

The classic application of sample and hold is to create something called a “stepped random control voltage”. By sampling white noise with a steady stream of pulse (preferably from a LFO) a “stepped random control voltage” is achieved, creating a sound famililar from robots and droids of the sci-fi flicks of the 1970ies. Another application is to trigger it at high frequency to create a bitcrusher (look at the preset "*Bitcrusher*" in the Modular FX for examples of this).

Slew Limiter

A slew limiter is a device that smoothes an incoming signal, limiting the maximum rate of change of the output voltage per unit of time. It is often called lag generator. It creates a *glide* or *portamento* when applied to a note CV signal.

Parameters

Time This knob sets the slew time from 0 to 10 seconds.

Inputs

In Input CV signal here is smoothed out.

Outputs

Out CV output of the SLEW LIMITER.

In Use

Can be used on **Note CV** to create portamento or glide effects. Random bursts of white noise can be filtered into slowly evolving random CV changes.

Divider

This module outputs a clock pulse for every *x* pulses it receives. Use for example the **/3** output to trigger an output pulse for each third incoming pulse.

Parameters

Divide	Sets the division of the clock on the **Out** output.

Inputs

In	Clock pulse input.
Reset	Reset pulse output and all internal counters and clocks.
CV	CV control the clock division of the **Out** jack.

Outputs

Out	This is the programmable division output. It outputs pulses divided between 2 and 32 times and are affected by the **Divide** offset knob and the **CV** division input.
/2	Outputs a pulse for every 2nd input pulse.
/3	Outputs a pulse for every 3rd input pulse
/4	Outputs a pulse for every 4th input pulse
/8	Outputs a pulse for every 8th input pulse
/16	Outputs a pulse for every 16th input pulse

In Use

Clock division means that clock pulses arriving at the input are counted and output pulse will be generated for example every 16th pulse at the /16 output. The division module is most commonly used as clock divider, but can of course also be used as a suboscillator if connected with a pulse output of an oscillator.

Logic Tool

This module is used to perform simple logical operations on DC signals where signals over 2.5V equals a logical high state (1) and below 2.5V equals a logical low state (0). The module has two inputs with inverting logical outputs. The two inputs are compared in logical function truth tables and each logical function has its own output.

Inputs

In A Signal input of channel A: the first operand.

In B Signal Input of channel B: the second operand.

Outputs

Inv A Inverted logical output of channel A. Output is high for inputs below 2.5 V, as indicated by the LED.

Inv B Inverted logical output of channel B. Output is high for inputs below 2.5 V, as indicated by the LED.

AND Output jack for logical AND function. This output is high when **both** channel A and channel B inputs are high.

NAND Inverted function of the AND output jack.

OR Output jack for logical OR function. This output is high when either channel A or channel B, or both inputs are high.

XOR Output jack for logical XOR (exclusive OR) function. This output is high only when either channel A or channel B input are high.

NOR Inverted function of the OR output jack.

In Use

To make a sequencer stop mid sequence, you can connect the DAW SYNCS **Run** signal to **A**, and **Gate Out** from the sequencer to **B**, and use the **AND** output to control an audio source. This means that both the **Gate Out** and **Run** must be active for **AND** output to become active.

Signal Tool

This module acts as a toolbox for continuous CV signals and does not differentiate between logic high and low like the LOGIC TOOL (see above).

Inputs

In A Signal input of channel A, the first operand.

In B Signal input of channel B, the second operand.

Outputs

Neg. A Negative output of channel A. An input of for example 2V will give an output of -2V.

Neg. B Negative output of channel B.

ABS A Absolute output of channel A. An input of 2V and a signal of -2V will give same output of 2V.

ABS B Absolute output of channel B.

Max Outputs the maximum of channel A and B. If channel A has 2.5V and channel B has 2V **Max** will output 2.5V.

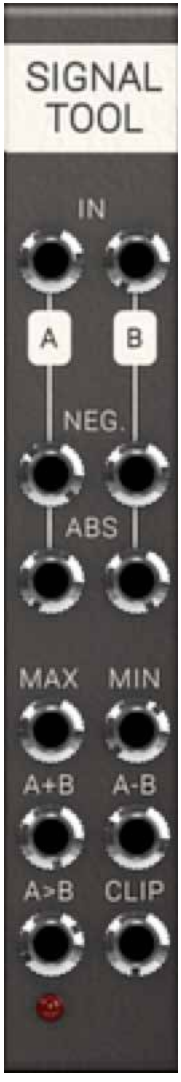
Min Outputs the minimum of channel A and B. If channel A has 2.5V and channel B has 2V **Min** will output 2V.

A+B Outputs the sum of channel A and B.

A-B Outputs the difference of channel A and B.

A>B Outputs logical high (5V) when A level exceeds the level of B.

Clip Clips channel A at channel B's input level.



In Use

The signal tool has many uses, for example, it can be used to compare levels on incoming audio on Modular FX to create faux compressor type patches.

Switch 1 to 4

This CV controllable switch can be used to direct a CV source or audio signal to any of four different destinations.

Parameters

Select	Select which of the four outputs that will be connected to the input.
LEDs	Indicate which of the four outputs is active.

Inputs

In	Input jack of the switch.
CV	CV input for controlling the connected output.

Outputs

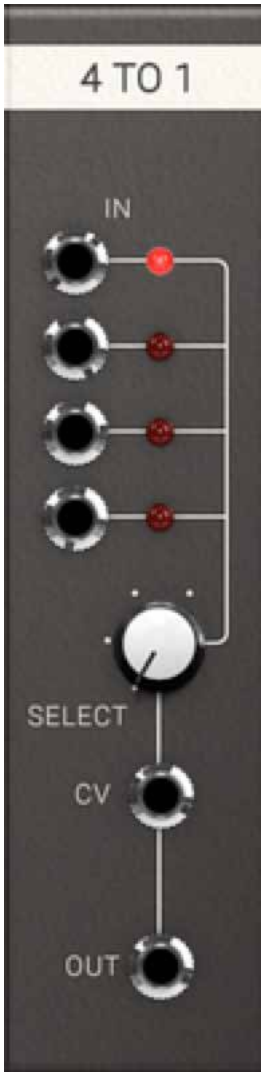
Out 1 - 4	Output jacks.

In Use

The switch 1 to 4 can have a lot of uses such as creating interesting modulation routes or pseudo-sequenced noises.

Switch 4 to 1

CV controllable switch that can be used to select one of the four connected CV or audio sources and send it to a set destination.



Parameters

LEDs	Indicate which of the four inputs is active.
Select	This knob is the offset for which of the four inputs will be connected to the output.

Inputs

In 1- 4	Input jacks of the switch.
CV	CV input for controlling the connected input to the output.

Outputs

Out	The output jacks of the Switch 4 to 1 module.

In Use

SWITCH 4 TO 1 can be used to select modulation routes or change distributed clock divisions for your sequencer. On some of the presets for the Modular FX, the SWITCH 4 TO 1 module is used to turn on and off an effect, guitar-pedal style.

Dual Pan

Two channel, CV controllable panning module that can be used to place a mono signal in a stereo mix, or to use as an A/B switch. The panning curve uses a constant power pan law with a +3 dB boost at the edges.

Parameters

[pan indicator LEDs]	Indicates the panning balance for the left and right outputs of channel 1 and channel 2.
Pan knob	Sets the **Pan** offset on its respective channel. The overall pan amount is determined by the **CV** input and the **Pan** knob.

Inputs

In	Audio input jack of channel 1 and channel 2.
CV	CV input for external control of the pan of channel 1 and channel 2.

Outputs

L, R	The outputs of channel 1 and 2.
1+2	These are the combined outputs of channel 1 and 2. L outputs from channel 1 and 2 are mixed together in output L 1+2, and R outputs from both channels are mixed together in output R 1+2.

Dual Pan in use

The DUAL PAN module can be used to manually or automatically pan a mono signal in a stereo mix. A typical example would be two percussive sounds that the user wants to place individually in the stereo image. The summed output, **1+2**, can then be used to add the two stereo panned signals together.

You can also use the DUAL PAN as an A/B switch to send a CV or audio signal to different inputs.

Dual X-Fade

Two channel, CV controllable crossfade module that can be used to mix two mono signals with a CV controllable balance input. Crossfade curve response is linear with a -6 dB dip in the middle.

Parameters

[crossfade indicator LEDs]	Indicates the crossfade balance for the LEFT and RIGHT inputs of channel 1 and channel 2
X-fade knob	Sets the crossfade offset on its respective channel. The overall crossfade amount is determined by the CV input and the **X-fade** knob.

Inputs

L, R	Input jacks for the two signals to which the crossfade is applied.
CV	CV input for external control of the crossfade function.

Outputs

Out	The outputs of channel 1 and 2.
1+2	This is the combined output of channel 1 and 2.

Dual X-Fade in use

The DUAL X-FADE module can be used for mixing different waveforms together, switching between sources, or modulating sources.

A typical use case would be an automated fade between two signals. Cross-fades can for be used as a pseudo-low pass filter when fading between outputs from the same oscillator, for example a sine or triangle on input 1, and saw or square on input 2. Then use an envelope or LFO to "sweep the filter" by patching the CV input.

Utility Quantizer

The Utility Quantizer module looks at input CV and outputs a quantized CV based on either a chromatic scale (top section) or a user definable scale (set on the bottom section). Chromatic scale means that each octave of 1V each is divided into 12 equal parts that corresponds to a chromatic 12 tone scale on a standard black and white keyboard.

Parameters

C to B buttons In the bottom section of the Quantizer the user definable scale is found. Each button active (on) represent a note part of your user definable scale.

Inputs

Chromatic CV in This is the CV input that you want quantized to a chromatic 12 note scale.

User Definable CV in CV input for the user definable scale part. Input any control voltage here that you want to be quantized to your custom scale.

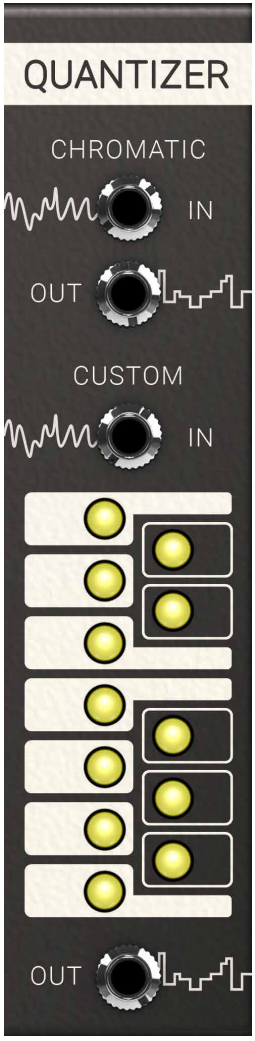
Outputs

Chromatic CV out This is the CV output of the 12 note chromatically quantized voltage.

User Definable CV out CV output for the user definable CV scale. The output CV here is quantized to the notes active via the C to B buttons.

In Use

The Utility Quantizer is great fun! Use it for example to extract musical material out of the random output of the A-118 Noise module. Or apply it to any kind of sequencer in order to stay in the right (customized) key when you transpose the sequence.



Heartbeat Modules

Heartbeat consists of seven drum modules, and a dedicated two-channel one knob equalizer. These are the drum modules from Softube Heartbeat, and if you own and have installed Softube Heartbeat the modules are available right from the Modular.

The drum modules are the same as in Heartbeat, with the addition of input and output jacks. If you need more information on each individual module, please see the Heartbeat chapter in this user's guide.

Heartbeat Bass Drum 1

The Bass Drum 1 channel from Heartbeat—the drum synth with a soul! It's very versatile and can sound anything from analog boomy to acoustic dry. All the parameters that are velocity controllable in Heartbeat can be CV controlled in the Modular.

Heartbeat Bass Drum 2

Circuit modeled from a classic Japanese drum machine from the early 80s. The parameters that arevelocity controllable in Heartbeat can be CV controlled in Modular.

Heartbeat Snare/Rimshot

A mixture of carefully selected waveforms and synthesis. It is designed to do snare drum and rim shot sounds. All parameters that are velocity controllable in Heartbeat can be CV controlled in Modular.

Heartbeat Snare/Clap

It is a mixture of carefully selected waveforms and synthesis. It is designed to do snare drum and clap sounds. All parameters that are velocity controllable in Heartbeat can be CV controlled in Modular.

Heartbeat Hihat

A mixture of carefully selected waveforms and synthesis. It does hihats, both analogue sounding and acoustic. All parameters that are velocity controllable in Heartbeat can be CV controlled in Modular.

Heartbeat Percussion

The percussion channels from Heartbeat.

Heartbeat Cymbal

Purely generated by modeled analog synthesis and draws inspiration from several early 80s Japanese drum machines. But the ring parameter has been added for the ability to get a more bell-like high pitched ringing sound. All parameters that are velocity controllable in Heartbeat can be CV controlled in Modular. Please refer to the Heartbeat User manual for tweaking examples.

Original Heartbeat instrument plug-in.

Heartbeat EQ

Each mixer channel in SOFTUBE HEARTBEAT has a one knob equalizer, with tailor made EQ curves for each drum sound. The close integration between these equalizers and its drum is a large part of how to shape the sound from the drum, but instead of adding the EQ to each drum module we decided to make it a module on its own so that you can use it wherever you feel it works.

Parameters

Drum E	Boosts or cuts the frequency range where the tonal content of the instrument selected by mode switch is found. The neutral setting is at 12 o'clock, turn counter-clockwise to cut this frequency range and clockwise to boost.
[mode]	Selects which frequency bands that the EQ affects. For example, BD boosts or cuts the low frequencies of the BASSDRUM and so on. OFF is bypass mode.

Inputs/Outputs

In	Audio input.
Out	Audio output.

In Use

Use the Heartbeat EQ it to equalize your Heartbeat drum sounds or why not add some bass to your favorite synth sound? Each setting (BD, SD RIM, SD C, HH, CYM, PERC) has its own sweet spot. OFF is bypass.

Credits

Oscar Öberg – concept, modeling, sound design. **Kristofer Ulfves** – concept, marketing, presets, user's guide. **Björn Rödseth, Kim Larsson** – modeling. **Arvid Rosén** – modeling and validation. **Patrik Holmström** – graphics programming. **Niklas Odelholm** – graphic design, programming, model validation. **Paul Shyrinskykh** – quality assurance. **Torsten Gatu** – programming. **Henrik Andersson Vogel** – marketing, user's guide. **Danjel van Tijn** – hardware design and model validation. **Dieter Doepfer** – hardware design and model validation. **Chris Assall** – model validation. **Kabuki** – presets.

34 Monoment Bass

It is not always possible to achieve a modern bass sound with "just a synth". While it is possible to create absolutely fabulous and punchy sounds with two oscillators, modulation and filters, a modern production usually requires more. And that "more" often consists of layers, movement, stereo imaging, and other effect processing that is normally done after recording the synth line.

But it doesn't have to be that way. In **Monoment Bass** we threw out the "ordinary" oscillators and replaced them with a collection of well-crafted stereo samples; taken from unique, highquality synths, processed through the finest boutique gear, and sometimes with layer upon layer to create the sources that you need to craft your sound.

On top of that we created a synth workflow with filters, envelopes and modulation, all easily accessible and adjustable, for you to shape the sources into the tone you need right now.

And if the textures of the sources aren't enough, there's an effects section with ambience, spatialization, distortion, multi-band compressor, and equalization. All that you need to make your bass line take the desired position in the mix.

It's about sound quality and workflow. A modern bass sound in no time.

User interface

1. Source section
2. Filter and LFO section
3. Effects section, Tone section

The user interface consists of four parts:

1. **Source section,** where you select different sound sources
2. **Filter and LFO section,** where you set the cut-off frequency of the filter, or use filter modulation to automatically change filter cut-off
3. **Tone,** where you balance your sources and set the punchiness of the sound.
4. **Effects,** the place for distortion, reverb, EQ, compression and spatialization.

On top of all this you also have a quick selector for presets that allows you to easily step through the presets in the Preset Collection, or if you've narrowed down the search by using tags or search words, step through the presets in the current search.

Quick preset selector

The Quick Preset selector shows the name of the current preset. If the current preset has been altered from its original state, its name is shown in italics.

You can easily step through the presets by using the arrow buttons. If you want more in-depth information on a single preset, click the preset name to call up the Preset Collection, in which you can read descriptions and usage information.

If you use the Preset Collection to search for presets, for example just showing presets with the tag "DISTORTED" or by searching for presets with the word "SUB" in the name or description, the Quick Preset window will remember the search and only step through the presets in the current search.

The Quick Preset window will indicate the status of the preset by using colors (white and gray) and font styles (normal and italic):

Normal font:
this is the current preset, and no parameters have been changed.

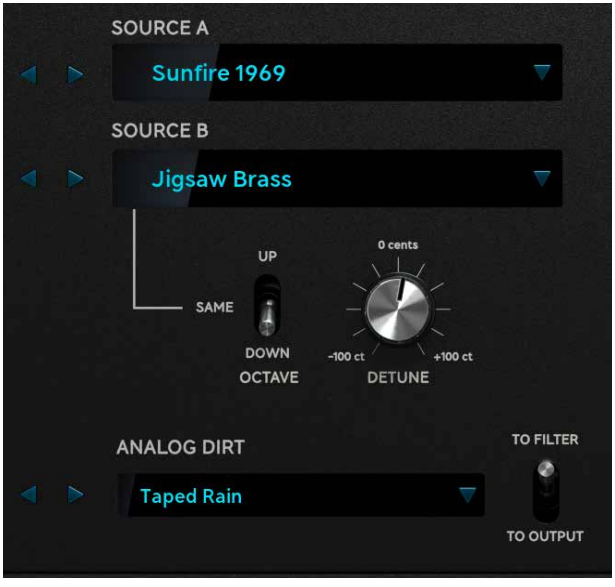
Italic font:
the current preset has been changed from its original setting.

Gray color:
the current preset is no longer part of the search results in Preset Collection.

Source section

The Source section's main purpose is to select the sources that are the foundation of the bass sound. You do that by clicking on the title of the source and choose which source you want from the different categories. On the second source (SOURCE B), you can also select which octave it should play in, so that you can have SOURCE A and SOURCE B playing an octave apart for an even fatter sound, or you can slightly de-tune SOURCE B to get a nice chorus-y effect.

The last source does not come from pristine and high-quality synths, but from the different flaws they have. We call it ANALOG DIRT and these are a unique collection of noises that have been carefully sampled and adapted for Monoment Bass. These can be used as an almost regular noise source, or can be filtered together with the other sources and together create complex textures and richness. There's just something about good noise. It just makes everything feel more alive! It's like a pretty black-and-white photo. The noise makes it real, and without it, it would just look like something you snapped with your mobile. But the noise has to be *right*.



The level of noise, and also the amount of analog "goodness" is controlled by the **Aging** control in the TONE section.

To set the mix between SOURCE A and SOURCE B, use the **Source Mix** knob in the TONE section.

Parameters

Source A, Source B	Selects the sound for the first or second source. Click on the < > arrows or use the dropdown menu to navigate the different categories.
Octave	Sets the octave (-1, 0, +1) for the second source.
Detune	Detunes the second source. A bit of detuning makes everyone happier!
Analog dirt	Sets the sound of the analog dirt. Choose between noise sources and different attack sounds.
Dirt to filter	Engage if you want the ANALOG DIRT to be filtered by the same filter as SOURCE A and B. Otherwise it will go directly to the EFFECTS section. The amount of ANALOG DIRT is effectively controlled by the **Aging** knob in the TONE section.

What source-ry is this?

The sources are divided into different categories, to help finding the right source quickly. Below we describe each category, and also point you to our favorite sources.

Analog Clean

A collection of rather clean analog sources without any harshness and distortion. These sources are a good starting point for add distortion using the DRIVE section. By combining a clean and less clean source you can create an unique sonic richness.

Listen to: **"Monster Saw"**, a rather fat analog sound designed with the Schmidt Synthesizer, try out DRIVE with the MODERN setting.

Analog dark

A collection of analog sources that don't have a full frequency spectrum and sound rather dark. Each source use specific filtering of the synthesizers they were sampled from, which gives another character using the filter section in Monoment Bass.

Listen to: **"Voltage Overload"**, sound designed using the Schmidt Synthesizer, using Schmidt's unique filters to darken the sound.

Analog Punchy

A collection of analog sources with a punchy attack, mainly designed with a filter envelope. Each synths' envelope and filter have something special and we didn't just want to offer oscillator sounds from several synths, but also capture the different filter and envelope behavior.

Listen to: **"Unisono puncher"**, sound designed using the Schmidt Synthesizer, but here we used the filters and their envelopes to create a punchy source.

Analog Rich

A collection of full spectrum analog sources. Don't call it super saw, these are all analog sources with analog richness!

Listen to: **"Big Ship"**, this monster sound was designed with the Modal 002 synth, play it again and again with open filter and Aging above 60% to hear the rich and lively sound.

Digital Clean

A collection of rather clean digital sources without any harshness and distortion - but with early digital "low resolution" fun! These sources are a good starting point for adding additional distortion. By combining a clean and less clean source you will get unique sonic richness.

Listen to: **"Early Digital 1"**, this sound was designed with the Modal 002 synth, capturing sonic fingerprints similar to the famous PPGs. Try out DRIVE with the MODERN setting.

Digital Dark

A collection of analog sources that don't have a full frequency spectrum and sound rather dark. Each source use specific filtering of the synthesizers they were sampled from, which gives another character using the filter section in Monoment Bass.

Listen to: **"Winter Night"**, a really warm, but digital, sound designed with a rare Jomox Sunsyn, which great filters we had to use!

Digital Noisy

A collection of rather noisy digital sources of all kinds. The different kinds of harshness was designed with on-board features of the specific synths like distortion, *FM* or other modulations, as well as with external processing of the sources.

Listen to: **"Truck FM Radio"**, a sound designed using the Nonlinear Labs C15 synth - now you can hear that Monoment Bass is not a standard bass synth, but an instrument with really unique and inspiring sample sources!

Digital Punchy

A collection of digital sources with a punchy attack, mainly designed with a filter envelope. Each synths' envelope and filter have something special and we didn't just want to offer oscillator sounds from several synths, but also capture the different filter and envelope behavior.

Listen to: **"Synclavier FM1"**, did we say filter envelope before? This sound was designed using the famous Synclavier synth. It uses no filters, but has incredible sound shaping features, like harmonic envelopes and more.

Digital Rich

A collection of full spectrum digital sources. For these sources you have to try out FILTER section and filter envelopes to form the rich spectrum in the right way.

Listen to: **"Outburst"**, a monster of sound. Designed with the Nonlinear Labs C15. Make sure to use the filter here, unless you want the audience to panic from all fatness!

Organ

A small collection of sources that remind of organ sounds. These sounds are a fundamental part of timeless dance and house tracks, but they definitely fit many other genres as well.

Listen to: **"Hot Washed Organ"**, another sound designed with the Nonlinear Labs C15, not a typical house organ, but a really deep and noisy organ for your next pop, dance or indie track!

Processed

A small collection of sources that have been heavily processed with analog filters, FM modulation, secret stomp boxes and hardware modular devices. These sources enrich the Monoment source collection with sonic content, far beyond standard waveforms.

Listen to: **"Circuit Board Fire"**, the name isn't entirely accurate, but that's what it sounds like. Another sound designed with the Nonlinear Labs C15 and several secret processing tools on top.

Sub

A collection of sub bass sources, typically with a low harmonic content, which is useful for creating sub or 808 like basses. Be careful with your speakers! These sources are very useful when you need to add more bottom to any other source.

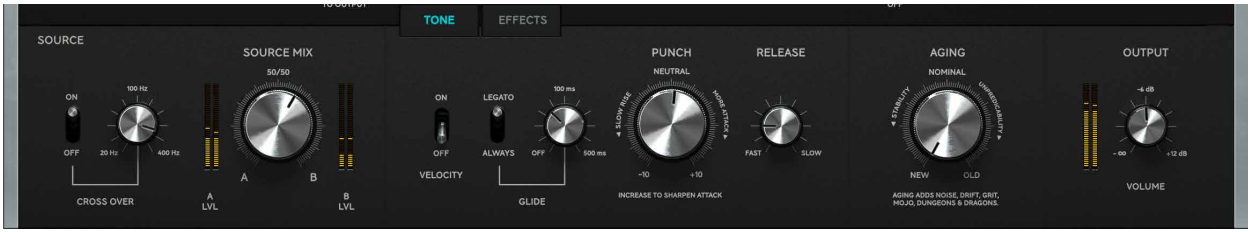
Listen to: **"Earth Shaker"**, the subtle richness and movement in the source makes even a single note bass line interesting. It was recorded with the Schmidt Synthesizer including filtering.

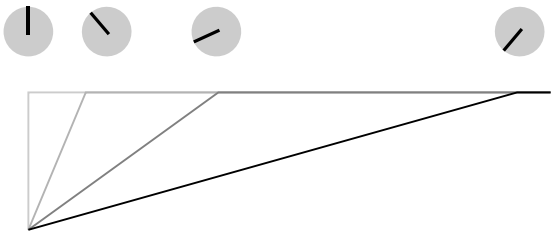
Tone section

The purpose of this section is to sculpt the tone: how you mix the sources, if it should have a sharp attack, or a slow fade-out. A lot of the "playability" of the sound can be adjusted here with just a few powerful controls.

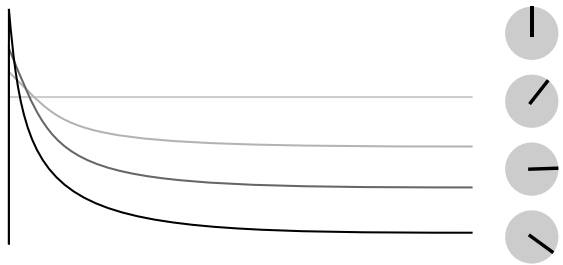
First step is to blend SOURCE A and B. In many cases it's enough with a single source, and you can set **Source Mix** all the way to A or B, but for modern production styles you often need to blend multiple sources. If you want to take the subs from SOURCE A, and the sizzle from SOURCE B, that's easy to do by engaging the **Crossover**. It separates SOURCE A and B at the **Crossover Frequency**.

Next step is to adjust the shape, the *envelope*, of the tone. That's easily done by adjusting the **Punch** control. Increase **Punch** and you'll get a snappier, punchier sound. Decrease it and you'll get a sound with a slow build-up. The **Release** sets the time it takes for the sound to fade-out.





Decreasing Punch from 12 o'clock down to nothing will make the onset of the note slower. This is useful for creating slow building sounds or to reduce the built-in attack of a source.



Increasing Punch from 12 o'clock will create a sharper attack with a lowered volume on the sustained note. Increase Punch to get a pluckier and sharper sound.

When designing Monoment Bass we wanted to make something that feels alive and real. Like a big beast that emits a lot of heat and creates earth-shattering lows. But sometimes you need something more static and a bit more predictable. That's what **Aging** does. More **Aging** gives you a more authentic, but also slightly more unpredictable sound with more dirt and grit. Don't want that? Fine, turn **Aging** all the way down.

Parameters

Crossover ON/OFF Activates the crossover filter between SOURCE A and B.

Crossover frequency Sets the crossover frequency. SOURCE A will get the low part, all frequencies below **Crossover Frequency**, and SOURCE B will get the frequencies above.

Source mix Sets the blend between SOURCE A and B.

Source meters Shows the volume level of SOURCE A and B after the crossover and mix.

Velocity ON/OFF Engage to get a velocity sensitive instrument.

Glide type Sets the type of glide or portamento.

Always on: the note's pitch will always glide between the former and the current note, in the time that is set by **Glide Time**.

Legato: the note will only glide when you play legato, ie. connected notes. If there is a space between to notes being played, the pitch will immediately jump to the latest played note. This mode is often called fingered glide, or auto glide.

Glide time The amount of time it takes for the glide to reach the target note. At its minimum position (0.0), the glide functionality is turned OFF.

Punch Sets the shape, the envelope, of the sound. At 12 o'clock it doesn't do anything. Increase it to get a sharper attack and more punch. Decrease it to get a softer attack with a slow build-up.

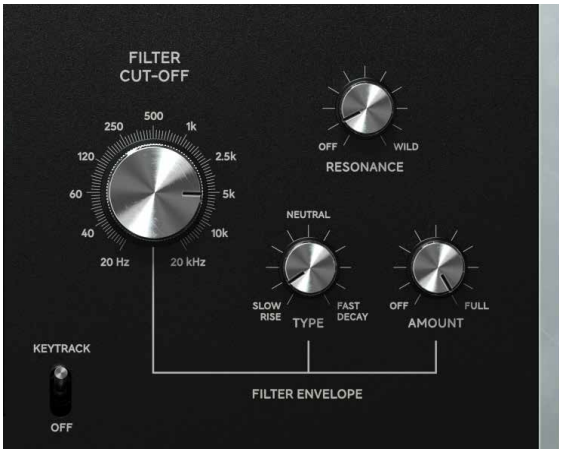
Release Sets the time it takes for the note to fade out.

Aging Sets the amount of unpredictability in Monoment Bass. More **Aging** means less stable pitch, more unpredictable attacks, more dirt and grit and more analog goodness.

Volume Sets the final output volume.

Output meter The level of the output signal.

Filter section



Even if a source, or a blend of sources, sound fantastic on its own, a synth bass sound is all about how you filter it, and how the filter changes when you hit a note. We want to make that as easy as possible and boiled it down to four main controls: the four knobs to the right.

Filter Cut-Off is the king. It rules them all. From subby bass to screeching high. That's what the **Cut-Off** do. **Resonance** is the trickster, that sets the amount of screechiness, and can be overdriven so that it distorts into a creative mayhem. (No, it's not your speakers breaking, it's how it should sound.) But the real heroes are **Filter Envelope Type** and **Amount**. They control how the filter changes when a note is being played! And that's how you get the classic synths sounds, like the "pluck" (the filter is closes when the note is played), or the "rise" (the filter is slowly opening then a note is played).

Filter Envelope Type goes from "slow rise" at its minimum position. The filter will slowly open up and reach the cut-off filter after a while. Turn it up to 11 o'clock and the filter will open up faster. At 12 o'clock it does nothing. Then when you increase it after that, the filter will close faster and faster, and at its maximum position the synth doesn't sound much more than a knock on your door. ("But hang on, I tried that, and IT DIDN'T DO NOTHING!")

Filter Envelope Amount controls how much of the **Type** effect you are getting ("Now you're telling me?") and with **Amount** on full, the filter sweep will go from nothing up to the cut-off frequency, or vice versa. Set it half-way, and the filter sweep will go from half-way between cutoff and nothing all the way up to the cut-off (or vice versa).

You can clearly see how the cut-off is set, how it is changing, and how much it can change in the filter graph.

1. Amount
2. Current frequency
3. Cut-off frequency

The combination of **Cut-off, Resonance, Type**, and **Amount** is by far the most powerful knobs in Monoment Bass. And we've spent a lot of time to make them as versatile as possible yet maintaining workflow and speed.

These filters distort. With too much resonance you'll get a very gritty (gritty-nice, not gritty-bad distortion!

Parameters

Filter cut-off Sets the cut-off frequency of the filter. Can also be adjusted by clicking and dragging the curve in the window.

Resonance Sets the amount of resonance (squealing) of the filter. With a lot of resonance, you can get the filter to distort in a very nice way! Can also be adjusted by clicking and dragging the curve in the window.

Filter envelope type Sets how the filter reacts to notes being played. Neutral at 12 o'clock, increase for a decaying sound, all the way to a short "plucky" sound. Decrease to get a rising filter sweep.

Filter envelope amount Sets how much the **Filter Envelope Type** should affect the filter cut-off. This is indicated by the shaded zone in the window. The current cut-off frequency (the combination of cut-off, filter envelopes and LFOs) can also be seen in the window.

In the window you can also set some additional filter parameters:

Filter ON/OFF Activates the filter.

24, 12, 6 Sets the type of filter:

24 dB/OCTAVE, classic analog synth filter with steep cut-off

12 dB/OCTAVE, typical 2-pole filter

6 dB/OCTAVE, a gentler filter used in a lot of modern productions. Technically, a 6 dB filter cannot be resonant, but many modern analog synths have combined the gentle 6 dB filter with a resonant circuit and achieving a filter like this.

You will notice that heavy filters (like 24 dB/OCTAVE) will remove too much of the built-in character that the sources have, while 6 dB/OCTAVE lets the sources keep more of their character.

LFO (low frequency oscillator) is another way of automatically change the filter frequency, but instead of having it change when a note triggers, you can get it to rhythmically change in time with your music. In general, you want to set it to sync the tempo with your DAW (SYNC ON) and set the **LFO Speed** to 1/4TH notes.

LFO ON/OFF Activates the filter LFO.

LFO Rise (The dot on the LFO Curve) Sets the start point of the LFO, from triangle to saw-tooth.

LFO Amount Sets how much the LFO should affect the filter cut-off, similar to **Filter Envelope Amount.**

LFO Shape Sets the shape of the LFO, from shark-fin to triangle.

Sync ON/OFF	Activates tempo synced LFO.
LFO Speed/LFO DAW Speed	Sets the speed of the LFO, in seconds in the first case, in beats/bar in the latter case.

A note on filter modulation: synth enthusiasts will probably scratch their head about how the modulation amount on the filters work, so let's state it clearly: the modulation will always lower the cut-off frequency. Set the cut-off to the highest you need and adjust amount until the modulation goes low enough.

Effects section

The EFFECTS section consists of five different effects that you can use to further sculpt your bass sound. These effects have been carefully chosen and designed for bass sounds, and the control set has been thoroughly tweaked to give you as much control as possible with as few knobs as possible.

When dialing in distortion on a bass sound it's often best to listen to the bass in context of the whole mix. Bass distortion sounds very different in a context of other high frequency instruments.

Drive

If you need more dirt and grit, **Drive** is king. For a typical bass sound you often need one of two different types of distortion: either a low frequency roar (TRANSF.) that adds harmonics to the lowest frequencies so that they can be heard even on a small mobile speaker, or a full-band distortion to make the bass sound sit better in the mix (MODERN).

The algorithms for the DRIVE section come from the Harmonics Analog Saturation Processor plug-in, so if you like them, or want more options for the distortion, please check out the Harmonics plug-in.

Parameters

Drive ON/OFF Activates the DRIVE section.

Type Sets the type of distortion,

TRANSF: a transformer based low-frequency distortion. Will only affect the lowest frequencies.

MODERN: a full-band distortion

Drive Sets the amount of distortion.

Ambience

Many have been taught that putting reverb on the bass is a classic mistake. That is often true, since the reverb smears out the bass in time, and you often need a very focused and tight bass. But what would creativity be if you can't break rules?

The ambience algorithms in Monoment Bass was designed for synth bass and are meant to enhance the stereo goodness and movement of the bass sounds. It can add amazing texture to your sound, and it is quite effective to use the SPATIALIZATION features in conjunction with the AMBIENCE to further tailor the sound.

Parameters

Ambience ON/OFF Activates the AMBIENCE section.

Tone Sets the tone of the AMBIENCE, with dark tones counter clockwise and brighter tones clockwise.

Dry/Wet Sets the amount of AMBIENCE.

Type Sets the type of AMBIENCE.

NEUTRAL: A short all-round reverb, created with a modern hardware device combined with analog filtering. This ambience gives your basses some natural "air" and size.

ELECTRIC: A short robotic reverb, created with a French reverb device from the 80s, combined with some secret offline processing of the recording. Use this ambience to create a very special sonic fingerprint of your basses.

METAL: A unique metallic reverb, created with a French reverb device from the 80s. This ambience gives your bass sound some industrial metallic sonic flavor.

RUSTY: A short unique reverb, created with a modern hardware device combined with offline pitch shifting and other vintage tools to give a rusty edge to your bass sound

DIRTY: A dirty reverb, created with a famous modular reverb device and combined with some secret stomp boxes. Use this ambience to create a trashy little space around your bass sound.

EQ

Equalization, tone control, is an essential part of all sounds. The EQ in Monoment Bass is an extremely powerful two-knob algorithm that, of course, has been tailored for bass synths (do we need to say that again?).

You can see the two controls (**Bass** and **Tilt**) as having two different purposes. With the **Tilt** knob you adjust the overall sound, the balance between the high frequency harmonics and the sub-frequency content. The **Bass** control is what you use to specifically target the bass frequencies. In other words: **Tilt** is used to shape the character, and **Bass** is for making it sit in the mix.

Parameters

EQ ON/OFF	Activates the EQ section.
Bass	Counter clock-wise: cuts bass frequencies Clock-wise: reduces high frequency content With the semi-resonant nature of the cut-filter it's very easy to find sweet-spots where the low frequency rumble has been removed while the fundamental frequencies are being emphasized. This technique has been used in many famous low-frequency EQ circuits.
Tilt	Turn clock-wise for a brighter sound, turn counter clock-wise for a darker sound. This is the knob you reach for when you need to balance the bass sound in the mix.

Multiband

The knob to rule them all. Multiband makes everything fatter, clearer, better, more defined… it's hard to not use it on everything. We got our product manager Paul, who also designed the preset algorithms for **Weiss MM-1** and **Drawmer S73**, to design a one-knob multiband compressor especially for bass synth, and the result is the multiband you see in Monoment Bass. With just a bit of compression you'll get a warmer and fuller sound. Dial it up even more and you get something that is ready to cut on a record. Overdrive it and it gets a character on its own.

Parameters

Multiband ON/OFF	Activates the multiband compressor.
Amount	Dial in the amount of compression.
Meters	From left to right: the amount of low-frequency gain reduction, the amount of mid-frequency gain reduction, the amount of high-frequency gain reduction.

Spatialization

All sources and algorithms in Monoment Bass are stereo, and sometimes you want to enhance the stereo-ness of them, or maybe reduce it. That's what the SPATIALIZATION block does.

The **Sub Mono** knob forces the audio below the set frequency to become mono, which is great if you're cutting vinyl or need a firmer low end. The **High Freq Widener** will increase the stereo information in the audio. When used in tandem, the **Sub Mono** will still make all frequencies below in mono, regardless of what the **High Freq Widener** is set at.

Parameters

Spatialization ON/OFF	Activates the SPATIALIZATION effect.
Sub Mono	All audio below the set frequency will be in mono. Pull it all the way up to force the output to be completely in mono.
High Freq. Widener	Increase to get more stereo width.

The SPATIALIZATION section is a mid/side matrix with built-in EQ. **Sub Mono** enhances the "mid" part while **High Freq. Widener** enhances the "side" part.

Sources of sources

We spent a huge effort selecting and recording the sources for Monoment Bass, in order to reach a new level of quality and fun using this product.

All samples were recorded in 96 kHz in stereo via Merging Technologies Hapi AD converters and resampled to 44.1 kHz after all external processing.

All samples were designed with and sampled from real hardware synths. We didn't sample the standard Moog or Roland synths for basses, but those exclusive and rare products such as the following synths:

- Schmidt Synthesizer
- Modal 002
- Non Linear Labs C15
- Kawai K5000
- Jomox Sunsyn
- Yamaha DX5
- NED Synclavier

Synths from four decades were sampled via secret processing chains of hand-picked boutique pre-amps, EQs and compressors. We sampled every 3rd note over two and a half octave and sampled every note three times to capture typical minimal sonic variations, controllable via the **Aging** knob.

The ATTACK samples in the ANALOG DIRT section are based on real recordings of objects with a certain attack sound, sonically fitting to a bass for additional punch and dirt. The NOISE samples are based on real noise source recordings and processed sounds from field recordings. Those noises add controllable dirt to the basses.

Credits

Tobias Menguser – Initial concept, multi sample sound recording. **Niklas Odelholm** – Sound, visual and product design, project management. **Björn Rödseth** – DSP programming. **Erik Sight** – Framework programming. **Filip Thunström** – GUI programming. **Jacopo Lovatello** – Filter design and modeling. **Paul Shyrinskykh** – Compressor sound design. **Johan Bremin** – Quality assurance. **Kristofer Ulfves** – Synth expert. **Ulf Ekelöf** – 3D graphics modeling. **Klaus Baetz** – programming. **Madison Mars, Vandalism, Cr2, WA Production, Black Octopus, Function Loops** – Factory presets.

MODULAR EXCLUSIVE

35 Mutable Instruments Braids

The Mutable instruments Braids eurorack module was released 2013 and has since then become one of the most desirable and popular modules among the eurorack crowd. It is a voltage controlled digital oscillator featuring 45 different oscillator algorithms in a wide variety of synthesis paradigms. Hidden beneath the surface is a plethora of useful functions such as built in quantization scales, envelope, VCA and bit reduction – Braids is truly the Swiss-army knife of oscillators!

Overview

1. Setup button
2. Down button
3. Up button
4. Edit dial
5. FM modulation knob
6. Timbre modulation knob
7. Color knob
8. Display
9. Coarse tune knob
10. Fine tune knob
11. Timbre knob
12. Trig input
13. V/Oct CV input
14. FM input
15. Timbre CV input
16. Color CV input
17. Output

Getting started with Braids

Braids is a very deep and versatile module, but here's some steps on how to get started and getting to know your Braids module in Softube Modular:

1. Select your Braids module from the module select view (brought up via the "Add" button on the center bar). When your Braids module is selected and placed in the virtual rack, connect Braids out jack to main out in order to hear what it sounds like. You will hear a buzzy sawtooth sound, the CSAW algorithm.

2. Now add MIDI control over pitch by adding a MIDI-to-CV module, found among the DAW and MIDI interfacing modules in the module select view. When this module is selected and added to your rack, connect the MIDI note output on the MIDI-to-CV module to the V/Oct CV input jack. Now, when your DAW plays MIDI notes, you will hear Braids module pitch tracking the notes played.

3. Braids features a multitude of different oscillator algorithms. Change the currently used algorithm by turning the edit knob (click on and drag your mouse up or down). Changing algorithms can also be achieved by clicking on the up or down button located on the top right on the Braids module.

4. Some algorithms in Braids need to be excited, pushed or plucked into oscillation in order to be heard. Change algorithm to no 31, "Bell". You'll notice that this algorithm is dead silent. Connect the trig output of the MIDI-to-CV module to the trig input of Braids. Now, when playing a MIDI note in your DAW, an inharmonic Bell sound is heard. You can change some properties of the bell sound by tweaking the timbre and color knobs.

5. Add a modulation source to perform some automatic tweaking of a parameter, for example by connecting an A-147 LFO to the timbre CV input of Braids. The modulation amount applied to timbre can now be set via the modulation "attenuverter" knob. Attenuverter means that no modulation is applied when this knob is at 12 o'clock, full positive modulation when it is fully clockwise, and of course full negative modulation when it is fully counter-clockwise.

6. Braids features a setup menu for deeper configuration. This mode is entered by clicking on the setup button. While in setup mode, this button remains lit to indicate that editing is being done within setup. Here you'll find settings for samplerate, bit-depth, quantisation, the internal vca and envelope. These features are all described in detail in the section below called "The Setup menu".

Braids oscillator algorithms overview

Mutable Instruments Braids features a formidable smorgasbord of different oscillator algorithms. There are 45 different oscillator algorithms, all listed below. The list below features a short description of each of the algorithms, what they are and what function the timbre and color knobs have.

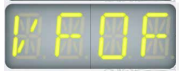
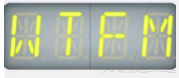
Classic analog waveforms

Wave	Description	Timbre	Color
	CS-80 imperfect saw	Notch width	Notch polarity
	Variable waveshape	Waveshape	Distortion/filter
	Classic Sawtooth/Square	Pulse width	Saw <-> Square
	Sine/triangle into wavefolder	Wavefolder amount	Sine <-> Triangle

Digital synthesis

Wave	Description	Timbre	Color
	2 detuned harmonic combs	Smoothness	Detune
	2 square VCOs with hardsync	VCO frequency ratio	VCO balance
	2 saw VCOs with hardsync	VCO frequency ratio	VCO balance
	Triple saw wave	Osc 2 detune	Osc 3 detune
	Triple square wave	Osc 2 detune	Osc 3 detune
	Triple triangle wave	Osc 2 detune	Osc 3 detune
	Triple sine wave	Osc 2 detune	Osc 3 detune
	3 ring-modulated sine waves	2/1 frequency ratio	3/1 frequency ratio
	Swarm of 7 sawtooth waves	Detune	High-pass filter
	Comb filtered sawtooth	Delay time	Negative/positive feedback
	Low-fi circuit bent sounds	Sample reduction	Bit toggling

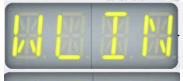
Vocal synthesis and formants

Wave	Description	Timbre	Color
	Direct synthesis of low-pass filtered waveform	Cutoff frequency	Waveshape
	Direct synthesis of peaking filtered waveform	Cutoff frequency	Waveshape
	Direct synthesis of band-pass filtered waveform	Cutoff frequency	Waveshape
	Direct synthesis of high-pass filtered waveform	Cutoff frequency	Waveshape
	Sawtooth with 2 formants	Formant frequency 1	Formant frequency 2
	Speaking-toy vowel synthesis	A, e, i, u, o	Gender
	Vowel synthesis	A, e, i, u, o	Gender
	A mixture of sine harmonics	The center frequency	Distribution of the amplitude of each harmonies
	2-operator frequency modulation synthesis	Modulation index	Frequency ratio
	2-operator frequency modulation synthesis with feedback	Modulation index	Frequency ratio
	Chaotic 2-operator frequency modulation synthesis	Modulation index	Frequency ratio

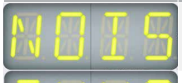
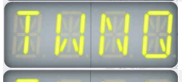
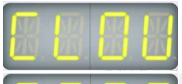
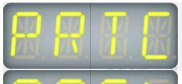
Physical simulations

Wave	Description	Timbre	Color
	Plucked string	Decay	Plucking position
	Bowed string	Friction	Bowing position
	Blown reed simulation	Air pressure	Instrument Geometry
	Flute simulation	Air pressure	Instrument Geometry
	Bell simulation	Decay	Inharmonic content
	Metallic drum simulation	Decay	Inharmonic content
	808 style kick-drum model	Decay	Click amount
	808 style cymbal noise spectrum	Filter	Squares <-> Noise
	808 style snare-drum model	Tone	Snappy

Wavetables

Wave	Description	Timbre	Color
	21 wavetables	Wavetable position	Wavetable selection (quantized)
	16 x 16 waves	X position	Y position
	Linear wavetable scanning	Wavetable position	Interpolation quality
	Quad wavetable scanning	Wavetable position	Chord type

Noise

Wave	Description	Timbre	Color
	Tuned noise (2-pole filter)	Filter Resonance	Response LP - HP
	Noise sent to 2 resonators	Resonance	Resonators frequency ratio
	Clocked digital noise	Cycle length	Quantization
	Sinusoidal granular synthesis	Grain density	Frequency dispersion
	Droplets granular synthesis	Grain density	Frequency dispersion
	Modem noises	Bit-rate	Modulated data

Parameters

Setup This button toggles between the Wave selection and Setup mode in Braids. When in Setup menu, this button is used to edit a value of a certain menu parameter (see further description below). You can always exit from the setup menu swiftly by clicking on the display.

Up arrow This button is equal to turning the edit dial one step upwards in a menu or increasing a value by one when editing a setup option.

Down arrow This button is equal to turning the edit dial one step downwards in a menu or increasing a value by one when editing a setup option.

Edit dial This dial scrolls upwards and downwards and is used to select between different Braids' different algorithms and menus.

Fine This is the fine tuning of Braids. Turning this knob tunes Braids one semitone up and down in pitch from its knob center position.

Coarse This knob controls the coarse tuning of Braids. It has a full range frequency that span over 9 octaves for most of the oscillator algorithms.

FM This knob is a so called attenuverter that controls the positive or negative frequency modulation of the currently selected oscillator algorithm. At counter-clockwise frequency modulation is at full minus 100%, at 12 o'clock 0% and at fully clockwise, it is at 100%.

Timbre The Timbre knob performs different functions for each oscillator algorithm. The value set is an offset to the Timbre modulation entered via the Timbre CV in jack which is being attenuated via the Modulation knob (see description below).

Modulation This knob is the amount of positive or negative modulation applied to the timbre parameter via the Timbre CV in jack.

Color The Color knob also performs different functions for each oscillator algorithm. Its value is also an offset to the control signals entered via the Color CV in jack (see further description below).

The setup menu

The Setup menu on Braids can be entered at any time by clicking on the Setup button when in Wave selection mode. The display on Braids will then show the text "WAVE" on screen, indicating that we have now entered the first page of the setup menu. While in the menu, the Setup button LED remains lit **green** while browsing between the different setup options, and changes to lit **red** when setup is pressed again to edit/change a parameter.

Selecting a setup option.

While editing a setup option, the setup button LED is lit **red**.

In the setup menu, the edit Dial and/or the up and down arrows are used to scroll between the different setup menu pages displaying the different setup parameters. The setup button is then used for entering and exiting edit of a certain setup parameter.

Exiting the setup menu is done by clicking the setup button again when on the "WAVE" screen (or on the "v 1.8" firmware version screen). But remember that **you can always also swiftly exit from the setup menu by clicking on the display** - this will send you directly to the currently used waveform page.

Listed below are all setup parameters:

WAVE	This is the "transition" page of the setup menu. If the setup button is clicked when on this page, setup menu is exited out to the Wave selection mode.
META	This parameter changes the behavior of the FM CV input if engaged. When META is set to "on", the FM CV input jack can be used to change between the different oscillator synthesis models.
BITS	This parameter sets the bit-depth of the data sent to the (emulated) DAC.
RATE	Selects the refresh rate of the (emulated) Braids DAC.
TSRC	This parameter selects a trigger source. "EXT." uses the gate/trigger input jack while "AUTO" option makes Braids detect changes in the V/Oct CV input larger than a semitone and generates trigger on each of these.

TDLY Trigger Delay. This parameter applies a delay between the moment the trigger is received and the moment the note is "struck" on the physical models.

|\ATT This is the attack time of the internal AD envelope generator.

|\DEC This is the decay time of the internal AD envelope generator.

|\FM This is the amount of FM modulation induced by the internal AD envelope generator when triggered.

|\TIM This is the amount of Timbre modulation induced by the internal AD envelope generator when triggered.

|\COL This is the amount of Color modulation induced by the internal AD envelope generator when triggered.

|\VCA This toggles the internal VCA on and off (default value). When set to "on", Braids will remain silent until the internal AD envelope generator is triggered as it will then control the VCA level.

RANG This parameter determines the range of the Coarse knob. The option "EXT. " adjust the range of this knob to +/- 4 octaves around the note received on the V/Oct CV input (if this option is selected a very low frequency will be heard if nothing is connected to V/Oct CV input). The option "FREE" (default value) adjust the range of the coarse knob to +/- 4 octaves centered around C3 (261.5 Hz). The optional "XTND" mode provides a larger frequency range but disables accurate V/Oct scaling as a side effect. The last option "440" locks the oscillator frequency to 440Hz exactly – helpful when tuning another VCO.

OCTV Octave selection. Transpose the oscillator up or down.

QNTZ Quantization of incoming cv on the V/Oct CV input jack. The frequency can be quantized to semitones, to any one of the other available scales or disabled (default value).

ROOT This setup parameter selects the root note for the selected quantizer scale.

FLAT When engaged, this parameter applies a detuning in the lower and higher frequencies to recreate the tuning imperfections of VCOs.

DRFT This parameter recreates the drifting of a badly designed VCO.

SIGN When engaged, this parameter applies grunge, glitches and imperfections to the output signal.

v 1.8 This is the last item in the setup menu and it shows the firmware version being emulated in Braids for Modular. Clicking on setup here also exits to the Wave selection menu.

Inputs

Trig in jack The trig in jack on Braids has three functions: Braids physical models need to be "excited" by an impulse on this input to give birth to a sound. Other models will treat the trigger as a reset signal, bringing the phase of the oscillator algorithm to 0. This envelope can also be used to trigger the internal AD envelope applied to the parameters of your choice in order to create sound animation without the need of an external envelope.

V/Oct CV in jack This is the 1V/Oct frequency input. Use this input to control the pitch of Braids from the MIDI-to-CV module or from a CV sequencer.

FM in This is a frequency modulation CV input. The Scale and polarity of the signal input here is controlled via the FM attenuverter knob.

Timbre This is the CV modulation input of the Timbre parameter of each oscillator model in Braids. The CV modulation that is input via the Timbre CV in jack is being attenuated via the Modulation attenuverter knob to add a positive or negative modulation to the offset value set by the Timbre knob on the panel.

Color This is the CV modulation input of the Color parameter of each oscillator model in Braids. The CV modulation that is input via the Color CV in jack is being added to the offset set via the Color knob on the panel.

Outputs

Out This is the main output jack for audio output. Connect this to main output of modular to hear the results of your tweaks.

MI Braids module in use

There are many fun uses for a module as versatile as Mutable Instruments Braids, but here's some examples:

Standalone Synth

By activating the VCA function among the setup functions in Braids, the internal VCA will linked to the internal envelope. This example shows how you to link up your Braids module after VCA is set to "on" and TSCR to "Exti".

Polyphonic Synth voice

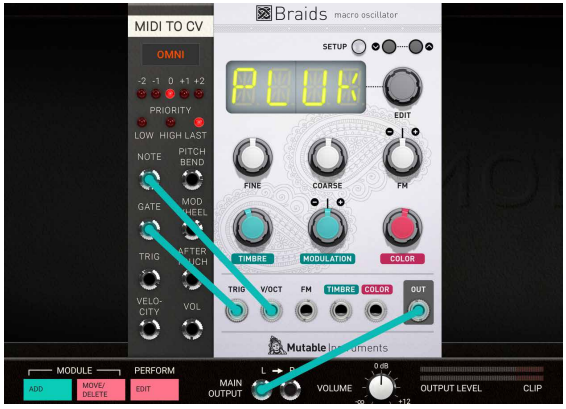
A similar technique that is described above can of course be used to create polyphonic voices (see example pictured above). Only downside to using the internal envelope of Braids is that it does not contain a sustain/ hold level and thus are not suitable for long sustained chords.

Wavetables

The oscillator algorithms "WTBL", "WMAP", "WLIN" and "WTx4" are all different takes on wavetable synthesis. This technique, used in some of the most iconic early digital synths, plays back parts of pre-recorded and processed single cycle waveforms. The timbre knob and CV input modulation set currently played back waveform. In the example pictured an LFO is sweeping the wavetable of a chord set in the "WTx4" oscillator mode.

Acoustic modelling

Using some the acoustic modelling oscillator algorithms such as "PLUK" can produce some really exciting results, especially in combination with some of the special microtonal quantisation modes such as "JAPA" or "GAME" that can be selected in the setup menu.



Drums

Braids has some very nice analog drum models for bassdrum, snare and cymbal-noise built in. In the example pictured the algorithms "KICK" and "SNAR" are triggered from the trigger-sequencer. A combination of the "CYMB" algorithm, a Doepfer A-132-3 DVCA and two envelopes are used to create open and closed hihat.

Wave sequencing

So called wave sequencing patches can also be creating with Braids by switching on the META function in the setup menu. This enables the user to create sequences where the CV for each step will control the algorithms currently used. In the example pictured, the META functionality is used to create a (monophonic) drum track where the CV knobs for each triggered step will determine which kind of drum will be played back as well as the pitch of that drum.

Credits

Eric Hampusgård – programming, modeling. **Émilie Gillet** – original code, feedback. **Kristofer Ulfves** – project management, presets, validation, user manual. **Oscar Öberg** – programming, mentoring. **Arvid Rosén** – mentoring. **Fanny Hökars** – user manual layout. **Bitplant** – GUI graphics.

MODULAR EXCLUSIVE

36 Mutable Instruments Clouds

Mutable Instruments Clouds is a granular audio processor. It creates textures and soundscapes by combining multiple overlapping, delayed, transposed and enveloped segments of sound taken from an audio recording buffer. The unique focus of Clouds is on real-time processing and manipulation of streaming audio, rather than playback of pre-recorded samples. This means that Clouds continuously records incoming audio through its inputs into the audio buffer, which can be up to 8 seconds long (by using the reduced audio quality setting). This limitation of the original hardware has been kept in the Softube Modular version in order to ensure a one-to-one hardware-software compatibility and similar user experience. The Softube Modular version of Clouds is based on firmware revision 1.31.

Using this continuously recorded audio, Clouds synthesizes a sonic texture by playing back short, overlapping segments of audio – also known as grains – extracted from it.

One could describe Clouds as a vessel with an audio-stream running through it, featuring a tap that drips audio droplets out of the vessel for us to listen to. The tap sets the pace at which the audio droplets are dripping out this vessel. The pace is however also controlled by the size of each drop.

Overview

1. LED meters
2. Audio quality selection button
3. Blend mode selection button
4. Pitch knob
5. Texture knob
6. Blend knob
7. Size CV input
8. V/Oct CV input
9. Blend CV input
10. Freeze push button
11. Size knob
12. Position knob
13. Density knob
14. Input gain knob
15. Position input
16. Trig input
17. Freeze gate input
18. Left and right audio inputs
19. Density CV input
20. Texture CV input
21. Left and right outputs

Freezing, fragmentation and dissolution of the instantly unexpected is the way of Clouds, rather than the careful planning of what might very well not come. Have fun!

Description

Getting started with Clouds:

1. A great way of understanding how Clouds works is to run a well known and easily recognisable piece of audio through it. Try for instance using a drum/instrument loop on an audio track in your DAW. Use Modular FX loaded as an audio insert effect on your track and route audio through Clouds in your virtual rack. You should now hear an eerie, ghost-like version of your audio loop playing through Modular FX and Clouds when you hit play in your DAW.

2. Try adjusting the Density knob of Clouds slowly from fully clockwise, via the 12 o'clock position, to fully counterclockwise while your loop is playing. Notice that the sound seems to "break up" and become silent around the 12 o'clock position, and then resume on the far side as you keep turning the knob. This is because Density controls the speed at which Clouds is playing back sample "grains" from the audio-stream that is being fed through the module.

7 o'clock to 12 o'clock positions on the Density knob plays back the audio at a regular pace which decreases the further towards the 12 o'clock position we get.

On the far side, 12 o'clock to 5 o'clock positions, the audio is played back at an irregular pace, starting slow around 12 o'clock and getting increasingly more rapid as we turn the Density knob towards the end-position, 5 o'clock.

3. Now try adjusting the Texture knob from the end position, fully clockwise, slowly over to fully counterclockwise. Notice how the sound character changes. This is because of the different volume envelope applied on each grain. While the envelope at fully clockwise (Hann window) is smooth, the envelopes at 12 o'clock (triangle) and at the fully counterclockwise position (square) sound more harsh. This is extra obvious when the Density knob is set to one of the slower rates (regular or irregular) around 12 o'clock.

4. Now try clicking on the Freeze button. It will light up green and the audio-stream through Clouds will remain frozen until the Freeze button is clicked again (or CV turns it off via the Freeze CV In jack; more on this later).

When Clouds is in its Freeze state, try "audio-scrubbing" through the frozen audio buffer with the Position knob. Find a start of a drumbeat or any other percussive sound in your frozen audio.

5. Now, reduce the pace at which the sampled "grains" are played back again by setting the Density knob to around 10 o'clock. You should now hear a steady pulse of audio grains played back. Try adjusting the Size knob. You'll notice that Size not only affects the length of the played back grain, but also the pace at which it is played back. Try also adjusting the Pitch shift knob. This shifts each grain up or down 2 octaves from the original pitch (at the 12 o'clock position).

6. OK. Next let's look at the Blend modes. Unfreeze Clouds by clicking on the Freeze button again and you will hear your DAW-loop playing through Modular FX and Clouds again. By clicking on the lower button to the right of the LED row you will cycle which Blend mode is active. Select the Blend mode furthest to the left with the drop symbol; the Dry/Wet mode. You'll notice that the Blend knob on the lower right on the Clouds module will change color to white and knob position is set to fully clockwise. This means that the Blend VC Dry/Wet parameter is currently set to fully wet mode. Tweak the Blend knob gradually to fully counter-clockwise to only hear the dry signal. Notice that the LED above the currently selected Blend mode changes color from unlit to green and then over yellow, to red at the Blend knob set to a fully clockwise position. The color will then always be reflected whenever tweaking the Blend knob regardless of which mode that is currently selected. In this way, you can always keep track of the four different Blend parameters current value when tweaking one. At VC Dry/Wet set to 12 o'clock you will hear both dry and wet signals at the same time and notice that there's a slight delay between the two. This is normal and the same behavior as in the original hardware. With the Pos knob you can set where in the audio buffer Clouds will playback of sampled grains, effectively creating a delay between recorded and played back audio. If a CV source is inserted to the Blend CV input, it will add to and control the currently selected Blend Mode parameter.

7. Let's move on to the next Blend mode, the VC Stereo Spread, indicated below the second LED from the left by a stereo image symbol. By repeatedly clicking on the Blend mode selection button until the second LED from the left is lit green, the VC Stereo Spread mode is selected. This is also indicated by the color of the Blend knob (now red) and the new position of the knob to reflect the current value of VC Stereo Spread. This parameter determines the stereo spread of each played back sampled grain, i.e. the stereo width of Clouds. Set the Blend knob all the way counterclockwise for fully mono output on the left and right outputs, or all the way clockwise for extreme stereo spread of each grain.
8. The third Blend mode, indicated below the third LED from the left by a curling arrow symbol, is VC Feedback. This parameter determines how much of the the output signal will be fed back into the input audio stream. Be careful with applying too much feedback as it is easy to get loud output from Clouds when the feedback starts building. When in Freeze mode this parameter changes to a kind of additional reverb control (for more information see detailed description below).
9. The last Blend mode is VC Reverberation. It is indicated below the fourth LED from the left by a box symbol and set how much internal reverb is applied to Clouds output.

Tip: Use performance module linking to gain easy access to all four Blend parameters at the same time.

Note: Since Clouds is essentially a digital module, it induces some latency to its processed signals. Time critical elements processed through Modular FX and Clouds will appear "out of sync" if not compensated for beforehand.

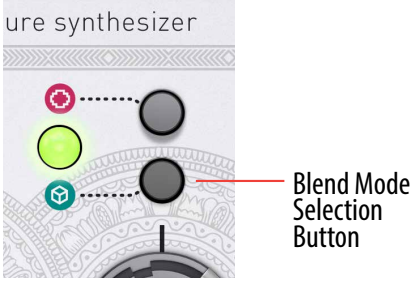
Parameters

Freeze Push-Button When this button is engaged (is lit), Clouds "freezes" the audio buffer held in memory and the current grain will continuously loop until the freeze button is dis-engaged again by pressing it a second time. During freeze all Clouds' controls will work as normal with all functions now applied to the grain "loop". Only two functions will be different: Input Gain will not affect the currently playing grain "loop" since it only controls incoming audio level. Also Feedback function will be given a different function since feedback can no longer occur in the recording buffer (because it is frozen). Instead, the output signal is routed through delays and all-pass filters to get the feedback build-up to occur in this extra recording space, giving the sound a very reverb-like nature.

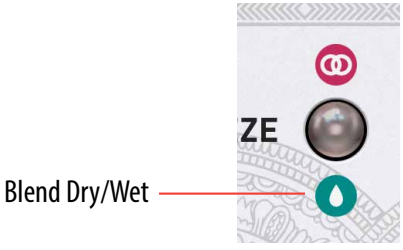
Audio Quality Selection Button This button toggles between the four different sound-qualities in Clouds: 16 bit stereo, 16 bit mono, 8 bit stereo and 8 bit mono. The current mode will be displayed briefly on the 4 LED bar by a red indication below the currently selected mode symbol (displayed above the 4 LED bar). Clicking on the rose symbols above the LED indicator bar will also swiftly change between each of the blend modes. Note that Clouds' 8-bit

has a lovely flavor of 8-bit μ-law companding which make it sounds like a cassette with less hiss, but more distortion.

Blend Mode Selection Button This button determines the current state of the Blend knob and input jack. Clicking on this button enables the user to toggle between (from left to right) VC Dry/wet mode, VC Stereo Spread, VC Feedback and VC Reverberation mode. The current mode will be displayed briefly on the 4 LED bar by a green indication above the currently selected mode symbol (displayed below the 4 LED bar). Clicking on the green symbols beneath the LED indicator bar will also swiftly change between each of the blend modes.



VC Dry/Wet Balance Mode When VC Dry/wet balance mode is selected, the Blend knob (and the input CV at the Blend input CV jack) sets the balance between the dry and wet signal, 'dry' being the unaffected portion of the sound fed through Clouds, and 'wet' being the affected signal.



VC Stereo Spread When VC Stereo Spread mode is selected, the Blend knob (and the input CV at the Blend input CV jack) will determine how each grain is distributed in stereo. In VC Stereo Mode, turning the Blend knob clockwise will "widen" the stereo output of the Clouds.

VC Feedback As VC Feedback mode is selected, the Blend knob (and CV input at the Blend jack) will set the internal feedback of Clouds. Depending on the settings of other parameters of Clouds such as Size, Density and Pitch, this feedback will be more or less harsh. Be careful though – it is quite easy to get Clouds feedback to self-oscillate which causes internal distortion. When the Freeze function is engaged, the

VC Feedback can no longer occur in the recording buffer (it is frozen) and thus feedback changes function to a kind of additional reverb control.

VC Reverberation In VC Reverberation mode the Blend knob (and the input CV at the Blend input CV jack) will control how much signal is fed into Clouds' built-in reverb. The Blend knob set fully clockwise in VC Reverberation mode will generate a very "wet" and long reverb tail.

Position Knob With the position knob, you will select from which part of the buffer the grains are taken, i e setting the starting-position from which the sample-grain are being played back in Clouds. If you change this knob while Clouds is processing sound that goes through, you will hear the played back audio affected in different ways. In Freeze mode, turning this knob will scrub the grain sample buffer back and forth while audio is played back.

Size Knob This knob sets the size of the played back grains. With this knob turned all the way counterclockwise, grains will be very tiny and sound metallic depending on what settings the density and texture knobs are set to. Turning the Size knob clockwise while sound is played back can result in cool "time-stretched" effects. Of course a motion similar to this can be automated through the size CV inputs.

Pitch Knob With the Pitch knob each played back grain can be played back at a different pitch and, by the nature of Clouds, the pitch-shifted signal will still retain its tempo and timing. Slightly pitch-shifting each grain is also extra apparent when using feedback. The fully counterclockwise position gives a minus 24 semi-notes (two octaves) down transposition, while fully clockwise transposes each grain 24 semi-notes (two octaves) up.

Input Gain Knob This knob sets the attenuation of the signal at Audio in jacks. Fully counterclockwise, signal will be completely silent. Watch the LED meters to monitor the input signal so that it's not distorting while being fed through Clouds.

Grain Density Knob This control sets Clouds internal playback density and behavior of the played back grains. If this knob is at 12 o'clock, nothing will be heard at all since the playback density of grains will be set to none. Twist the knob slowly fully counterclockwise to get more dense, regularly played back grains with a constant, high rate. Or twist the knob slowly fully clockwise to increase density but with irregular playback, which sounds different and smoother than constant rate.

Texture Knob This knob sets each audio grains volume envelope. In the fully counterclockwise position, the envelope curve is very step and will give a harsh sound to Clouds, especially when the Density knob is set high (fully clockwise or counterclockwise). At twelve o'clock, the Texture knob gives each grain a more sloped and softer triangular shape and at the fully clockwise position a Hann window shape – a raised cosine distribution is used. The latter gives Clouds a blurry, pleasant sounding texture when played back.

Blend Mode Knob This knob controls one at a time, the parameters determined by Blend mode button selection: VC Dry/wet balance, VC Stereo Spread, VC Feedback or VC Reverberation (see full description above at Blend Mode selection).

Indicators

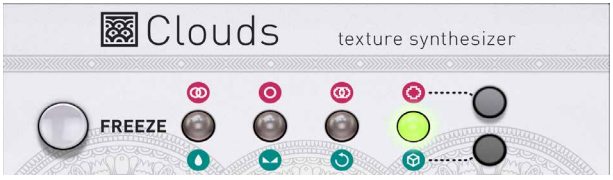
Freeze Button LED When the Freeze Button LED is lit, Clouds does not stream audio through it, but merely plays back what's currently in the audio-buffer.

Meters (4 multicolor LED bar) This bar of four multi-colored LEDs have different display functions. By default, it displays input level by lighting one LED after another from left to right. Adjust your input level by turning the In Gain knob until two or three LEDs remains more or less constantly lit (on sounds with an even sound level).

When the Audio Quality Selection button is clicked the LED meter bar will briefly show which audio quality is selected by a red LED indication. Red LED meaning (from left to right): 16 bit stereo, 16 bit mono, 8 bit stereo and 8 bit mono.

When the Blend Mode Selection button is clicked in order to change Blend mode, the LED meter bar will briefly show which mode is selected at the moment by a green LED indication.

When the Blend knob is moved, the LED at the position of the selected parameter will change color from none to green, over via yellow to red to indicate increased parameter value. The current values of the other Blend values will also be reflected by their LED colors at the same time.



Inputs

Freeze Gate Input Jack When a high (+5V DC signal) appears on the Freeze Gate input jack, Clouds will "freeze" the audio buffer and hold it in memory until the input jack goes low again. If the freeze button is engaged at the same time, it's current state will override the input CV of this jack.

Trig Input Jack When a trigger appears at the Trig input jack Clouds will trigger and play a grain, regardless of the density setting. This means that it is possible to trigger a short snippet of a "sampled" sound that is currently in the buffer while using another CV to sweep through the position in the buffer (see below).

Position CV Input Jack CV input into this jack mirrors the function of the Position knob. This means that you sweep the playback position of the audio buffer memory in real-time while audio is streaming through, or when in "freeze mode".

Size CV Input Jack This jack controls the played back size of the sample grains and mirrors that of the Size knob. Very cool time-stretch effects can be achieved by CV modulating this input, changing the parameter during playback

V/Oct CV Input Jack CV input jack to control the pitch transposition at 1 Volt per octave. This jack adds to the function of the Pitch knob, which works as an offset in relation.

Blend CV Input Jack (Controls The Currently Selected Blend Parameter) This jack controls the currently selected Blend parameter: VC Dry/wet mode, VC Stereo Spread, VC Feedback or VC Reverberation mode.

Audio In Left Left input jack for Stereo audio input.

Audio In Right Right input jack for Stereo audio input. This jack is normalized to Left input if no cable is inserted here (i.e. mono in).

Density CV Input Jack CV input jack to mirror the Density knob control.

Texture CV Input Jack CV input jack that mirrors the Texture knob control.

Outputs

Audio Out Left Left output jack for Stereo audio output.

Audio Out Right Right output jack for Stereo audio output.

The Clouds module in use

The Clouds module can be used in many interesting ways. Here's some suggestions:

Sampler

One of the coolest uses of Clouds in Modular is to use it as a kind of crude sampler. Use Modular FX to record your prefered sound into the sample buffer, freeze, set density to 50%(12 o'clock) and use the trig input to trigger individual grains. Use the Pos knob to scrub the buffer back and forth, and find the part of the Sample you want to play back. By adjusting the size from fully counterclockwise (only one sample grain played back) to full clockwise, the whole sample buffer can be played back from the start position set by the pos knob.

AM – RM kind of effect

By setting the Density knob at roughly 9 o'clock and Texture knob fully counterclockwise, the signal through Clouds take on a gritty tone. Now adjust the pitch knob to some odd ratio for clangorous rich ring-mod type sounds.

Delay

Using Clouds with full regular Density and Texture at fairly square (both knobs at fully counterclockwise), it can be used for some fairly straightforward delay-duties. Adjusting Size changes delay time and adjusting Blend feedback - you guessed it - delay feedback. Set the balance between the dry signal and the delayed signal with Blend Dry/Wet.

Pitch-shifter

This is a bit trickier. Since Clouds is not specifically made for pitch-shifting, different harmonic material requires slightly different Texture and grain Size settings for optimal pitch shifting duties. This is also depending on the ratio required - it is easier to achieve perfect one octave up or down pitch shifting with Clouds than for example a clean +7 semitones pitch shift. Experimentation is the key here.

Time-stretch

By modulating the size CV input with an lo frequency rising sawtooth wave with roughly the same speed as the played back audio through Clouds, a time-stretch kind of effect can be achieved.

Robotize

By automatically trigger freeze Clouds audio buffer, a cool "robotizer" effect can be constructed. This works exceptionally well on beats and rhythm material.

Credits

Eric Hampusgård – programming, modeling. **Émilie Gillet** – original code, feedback. **Kristofer Ulfves** – project management, presets, validation, user manual. **Oscar Öberg** – mentoring. **Arvid Rosén** – programming, mentoring. **Igor Miná** – user manual layout, hardware photos. **Bitplant** – GUI graphics. **Tomas Bodén** – presets. **Robin Rimbaud** – presets.

MODULAR EXCLUSIVE

37 Mutable Instruments Rings

The Mutable instruments Rings is a eurorack modular module that was originally released during winter 2015/2016. Rings brings physical modeling synthesis to Eurorack from a more modular angle than, for instance, the models featured in Braids and other modules from Mutable Instruments. Instead of trying to be a complete instrument, Rings focuses on the key ingredient, the resonator, ready to be excited by envelope clicks, trigger pulses, granular noise or any other audio source. The Softube Modular version of Mutable Instruments Rings is based on firmware revision 1.1 with the all features described in the original hardware's user manual.

User interface

1. Polyphony select button
2. Polyphony mode LED
3. Resonator mode LED
4. Resonator select button
5. Structure knob
6. Position knob
7. Position CV Modulation knob
8. Structure CV Modulation knob
9. Position CV input
10. Structure CV input
11. Frequency knob
12. Damping knob
13. Brightness knob
14. Brightness CV Modulation knob
15. Frequency CV Modulation knob
16. Damping CV Modulation knob
17. Brightness CV input
18. Frequency CV input
19. Damping CV input
20. Strum input
21. V/Oct input
22. In
23. Odd output
24. Even output

Rings is a resonator, the essential ingredient at the core of several physical modeling techniques. It transforms an external, un-pitched excitation audio signal (such as a click, a burst of noise, or whatever is fed into it) into a full-bodied pitched sound. You could describe Rings as the bar, the tube or the bunch of strings you'd cause to vibrate with an external signal in order to create your sound. However, Rings has some smartness already built into it that makes it very easy to use (more of this later).

Getting started with Rings

1. The easiest way to use rings is to connect it as a regular oscillator (well, almost). Add a MIDI-to-CV module and Rings to your rack.

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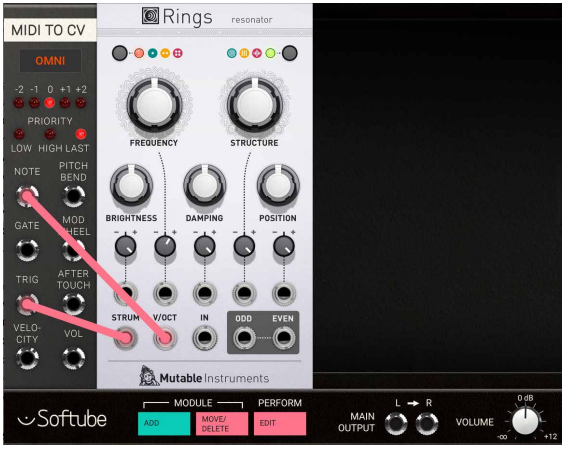
2. Connect the Note CV output of the MIDI-to-CV module to the V/Oct CV input on Rings.



3. Now connect the Trig output of the MIDI-to-CV module to the Strum input on Rings.

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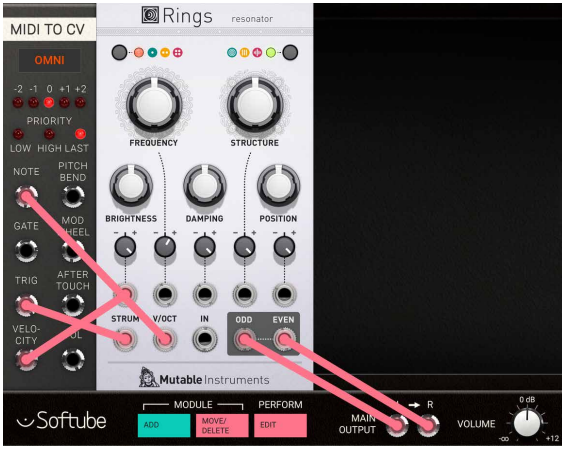
4. And now connect one of Rings' outputs (Odd or Even) to Modular's left main output. Now, the MIDI notes from your DAW should be able to play notes on Rings. Experiment by selecting between the three different internal models by clicking on the upper right button. The upper right LED color will change from green to yellow, and to red, as you change models and sound of Rings.

5. You can now experiment by adding a connection between the Velocity output of the MIDI-to-CV module to Rings Brightness CV input. Now, playing harder on your MIDI keyboard (or sending MIDI note information with less velocity) will make Rings sound brighter, while playing softer makes Rings sound will a duller/ softer sound.



6. By connecting the remaining Rings output (odd or even) to Modular's main right output, you will be able to clearly hear the dynamic panning effect between Rings' voices in polyphonic mode (top left LED mode red or yellow) and pitch spread in monophonic mode (top left LED mode green).

Interface

As mentioned earlier, Rings has some smartness already built into it that makes it easier to use without having to connect every input. The general design philosophy for Rings is that whenever you don't patch an input, the module tries to infer it from the other inputs. For example, the Strum trigger input indicates string changes, but if you don't provide it, the module will look at sudden changes in the
V/Oct CV input to decide if a new string is touched or not. If there's nothing in V/Oct CV input, Rings will detect transients on the In input. Similarly, if nothing is provided in the In audio input, the module will makes its own excitation signal by waiting for triggers in the Strum input, or, if there's nothing in this input, by detecting sudden changes in the
V/Oct input.

So, to sum it up: Ideally, **Rings** would need three input signals:

1. A trigger signal in the **STRUM** input, to indicate when the currently playing note should fade away, and when a new note is starting.
2. A CV signal in the **V/OCT** input, to control the note frequency.
3. An audio signal in the **IN** input, which will hit, strike or caress the resonator. But because it might not always be possible to get these three signals from your system, **Rings** makes the following assumptions:

- If no patch cable is inserted in the **IN** audio input, the module will synthesize its own excitation signal whenever a note is strummed. This excitation signal is either a low-pass altered pulse, or a burst of noise depending on the resonator type.
- If no patch cable is inserted in the **STRUM** audio in- put, the module will determine that a new string should be strummed either by detecting note changes on the **V/OCT** input. Or by detecting sharp transients of the **IN** audio signal when nothing is patched in the **V/OCT** input.

Rings Resonator models

The resonator select button, found at the top right on Rings, selects between the three different types of resonator models available. These are indicated by different LED colors on the top right LED and are:

1. Modal resonator (top right LED lit green)

Modal synthesis works by simulating the phenomena of resonance at play in vibrating structures. For instance, a string or plate will absorb certain frequencies while it will "ring" at some other frequencies, called the modes. Various materials or structures are characterized by different relationships between the frequencies of their modes, which Rings recreates. When we pluck a string, strike a drum or blow in a tube, the short burst of energy of the blow/impact contains many frequencies. Some of these frequencies will fall outside of the modes and are absorbed, while some of them will excite the modes and produces a stable, pitched sound. Each mode corresponds to a harmonic or partial in the spectrum of the sound, and is modeled by a band-pass filter. The Q factor of the filter will determine the sustain of the oscillations for the corresponding partials.

2. Sympathetic strings (top right LED lit orange)

Some string instruments, such as the sitar or sarod, make use of strings that are not directly struck/plucked by the musician. These strings are responding to vibration of the other strings, adding extra overtones or undertones to it. Rings simulates this phenomenon with a bunch of virtual strings (made with comb filters), allowing the addition of extra tones to an incoming audio signal. The tuning ratio between these strings can be altered.

3. Modulated/inharmonic string (top right LED lit red)

This last method is based on the extended Karplus-Strong method: the excitation signal is sent to a comb filter with an absorption filter, simulating the multiple reflection of a wave propagating on a string and being absorbed at its ends. However, to bring more variety to the sound, Rings adds three extra ingredients to this classic: a delay-compensated all-pole absorption filter creating more drastic plucking effects, delay time modulation emulating the sound of instruments with a curved bridge (like the sitar or tanpura), and all-pass filters in the delay loop, shifting the position of the partials and recreating the tension of piano string or completely bonkers inharmonic timbres.

Parameters

Polyphony select button This mode select button switches between Rings monophonic mode (LED lit green), duophonic mode (LED lit orange) and quadraphonic mode (LED lit red). Polyphonic means that a note that has been struck can will still be sounding when the next note is struck.

Resonator select button This button selects between the three different resonator models available within Rings (see detailed description above).

Frequency knob This knob controls the overall tune of Rings. This knob is unquantized when nothing is patched to the V/Oct CV input. However, when cable is connected to the V/Oct input, this knob will work as a quantized pitch offset instead.

Structure knob This knob controls the selected model's inner harmonic structure. Exactly how this is done differs slightly between the three different models: For the modal resonator (right LED lit red) the structure knob controls the frequency relationship between the partials and thus the perceived structure (plate, bar, string). When using the sympathetic string resonator mode, this knob will alter between set frequency ratios between the strings. And finally, when using the modulated/inharmonic strings resonator, this knob controls amount of modulation and detuning between the partials.

Brightness knob This knob controls the upper harmonics response of the excited signal within Rings. Low values on this knob simulate materials like wood and nylon, while high values simulate materials like glass or steel.

Damping knob The damping knob controls the decay time of the sound with a range between less than a 100ms set at the lowest value, up to about 10s, set at the highest value.

Position knob This knob controls the excitation position, meaning the point at which the string/surface where the excitation is applied. Applying the excitation right in the middle of the surface will cause, by summing, the even harmonics to cancel each other. This can be clearly heard when position knob is at 12 o'clock at Modal mode (right LED lit green) and in monophonic mode, no even output will be heard at all, as it is cancelled out.

Brightness CV Modulation knob This is the attenuverter for the Brightness CV input. This means that this knob controls the amount of positive or negative modulation applied to the brightness parameter via the Brightness CV in jack.

Frequency CV Modulation knob This is the attenuverter for the Frequency CV input. This means that this knob controls the amount of positive or negative modulation applied to the frequency parameter via the Frequency CV in jack. Note that the Frequency CV in jack is normalized to 3V when not connected, and thus will this knob work as a offset or fine-tune as long as no signal is connected to this jack (see also description below).

Damping CV Modulation knob This is the attenuverter for the Damping CV input. This means that this knob controls the amount of positive or negative modulation applied to the damping parameter via the Damping CV in jack.

Structure CV Modulation knob The attenuverter for the Structure CV input. This means that this knob controls the amount of positive or negative modulation applied to the structure parameter via the Structure CV in jack.

Position CV Modulation knob This is the attenuverter for the Position CV input. This means that this knob controls the amount of positive or negative modulation applied to the position parameter via the Position CV in jack.

Inputs

Brightness CV input This jack is for remote controlling of brightness within Rings. Control signals (CV) input here will affect the brightness parameter as described above.

Frequency CV input Signals input here affects the tuning of Rings. Note that the Frequency CV in jack is normalized to 3V when not connected.

Damping CV input This jack is for remote controlling of damping within Rings. Control signals (CV) input here will affect the damping parameter.

Structure CV input Remote control of structure. Control signals (CV) input here will affect the structure parameter.

Position CV input Signals input at this jack controls of the strike position within Rings. Control signals (CV) input here are added to the Position parameter.

Strum input This is the trigger input for polyphonic operation. A trigger received by Rings on this jack will assign a new voice and thus making "overlapping" voice-operation a reality. Try this out by setting the module in 4-voice mode (left LED lit red), turn damping fully CW to get super long releases, and just play your notes on the V/OCT input - they will overlap without cutting each other. A new trigger will cut the least recently played note and will start a new note – just like voice-stealing on poly-synths. When no cable is patched into this jack it is normalized to the V/Oct and In jack.

V/Oct input CV input jack for chromatic scaling (1V per Octave) for Rings. Connect your the Note CV from the MIDI-to-CV module here.

In This is the main audio signal input for excitation signal in Rings. If no signal is connected, this jack is normalized to an internal noise/burst generator that reacts to changes in the V/Oct input.

Outputs

Odd Audio output for odd harmonics. This jack is normalized to the Even jack if it is not connected, meaning that its audio will be mixed and output through there.

Even Audio output for Even harmonics. This jack is normalized to the Odd jack if it is not connected, meaning that its audio will be mixed and output through there.

The Rings module in use

Mutable instruments Rings is a very versatile and characteristic sounding module. Here's some uses:

An obvious use for Rings is to utilize the sympathetic string model (yellow right LED mode) in order to create credible plucked string sounds. Experiment with different brightness, damping, position and structure settings for different string sounds. Try also mapping velocity to brightness for additional expressive control over MIDI. An interesting exercise is also to replace the internal excitation signal with an external source for a different sound - the easiest way to try it in this patch is to add an additional patch cable from the trig jack out of the MIDI-to-CV module to the In audio input on Rings.

Using a random noise source or another continuous signal for excitation is a great way of creating rich moving drone-sounds. In this example we're using the modulated/inharmonic string model (red right LED mode) to create a deep drone.

Rings is great also for processing external sound! Try feeding any percussive, harmonic or non-harmonic sound into Rings using the external inputs of Modular FX to add flavor. Try for instance using the sympathetic string resonance model to create Sitar-esque drones for your guitar. Or why not adding cool modal resonating drones triggered from your drums?

Try using an LFO to slowly modulate the position of the modal model (top right LED green) for a cool phasing type of clock sound.

Make use of the polyphonic feature of Rings to create a guitar strumming emulation. In this patch we use an AD-envelope generator patched through a quantizer further on into Rings to create automatically strummed arpeggios in a selected scale. The left CV mix module limits the range of the envelope into the quantizer, while the right CV mix module adds together the arpeggio CV and note CV from the MIDI-to-CV module for transposition.

Create flute-like sounds by feeding enveloped white noise through Rings in the Modulated/inharmonic string model (top right LED lit red) with the resonance frequency tunes fairly high.

Credits

Eric Hampusgård – programming, modeling. **Émilie Gillet** – original code, feedback. **Kristofer Ulfves** – project management, presets, validation, user manual. **Oscar Öberg** – programming, mentoring. **Arvid Rosén** – mentoring. **Fanny Hökars** – user manual layout. **Igor Miná** - hardware photos. **Bitplant** – GUI graphics.

38 Mutronics Mutator

THE MUTRONICS MUTATOR WAS AN analog hardware unit that gained a lot of popularity among musicians, music producers and recording engineers in the 1990s and early 2000s. It can clearly be heard on records that had a big part in defining that era, with artists such as **Radiohead**, **U2**, **Nine Inch Nails**, **Depeche Mode**, **Chemical Brothers** and **Daft Punk** among its users. Production of the hardware ceased in 2007 since a vital component was discontinued by the component manufacturer, and it has only been available in the second hand market since then.

In the development work with the Mutator plug-in, Softube has worked closely with Mutronics to ensure that every little nuance of the original unit is faithfully captured. But we also made a few modern additions that makes the plug-in even more useful.

Overview

The Mutator is a stereo analog filter and envelope follower with full control facilities. It contains two independent voltage controlled filters, similar to those found in analog synthesizers, which can be used to treat any external sound source. Each filter can be controlled from its own associated low frequency oscillator (LFO) and/or its own envelope follower section. The envelope follower essentially controls the filter and/or volume based on the amplitude of the incoming (internal) sound source, or on an external sound source that is fed into the unit's sidechain. This external signal can be any audio source, eg. a drum sound, a guitar, a synth, a sampler output, or even a microphone. Thus the envelope characteristics of one sound can be superimposed onto the filtering contour of another.

Another switch on the envelope follower selects envelope follow mode or gate mode—in which the circuit detects whether a signal is above a certain threshold level before switching on.

In gate mode, the device can operate much like a dedicated noise gate unit, with the added capabilities of the analog filters. The LFOs have four different sweep waveforms each and may be switched via a stereo link to produce stereo panning effects.

The Mutator has two channels. The upper row of buttons control the left channel, the second row controls the right channel. It is possible to synchronize the LFO of the left to the right channel, as well as to the DAW host's song tempo.

The lowest row of knobs are additions Softube made to the plug-in, that were not part of the analog hardware. These include tempo syncing to the DAW host and a Mix button that lets the user blend the original sound with the mutated.

Knobs and Switches

The front panel controls for the Mutator are identical for both channels, with the exception of the **Link** switch which is detailed below.

Envelope Follower

Env Source Selects which signal drives the envelope follower circuit. In INT (internal) position, the sound source is also used as the control signal for the envelope follower. So with the **Env Source** switch set to INT, the filter section reacts to the amplitude of the sound source itself. In EXT (external) position, the envelope curve is determined by the incoming sidechain signal.

Env/Gate In ENV (envelope) mode, the envelope follower acts as its name implies, precisely following the amplitude of the control signal and producing a rapidly varying control signal which can be used to sweep the cut off frequency of the filter and/or volume level of the VCA. In GATE mode, the envelope follower is either full on or completely off, depending on whether the control signal is above or below the gate threshold as set by the sensitivity control.

1. Envelope follower
2. LFO
3. Filter section
4. Left channel
5. Right channel
6. LFO tempo sync
7. Output section

Sensitivity In Env mode, this sets the optimum drive level for the envelope follower circuit. Too low and the **Envelope LED** only glows faintly, too high and the LED is driven hard. The optimum setting is when the LED peaks in brightness at the loudest points of the control signal. In GATE mode the sensitivity control sets the threshold level at which the gate operates.

Attack This adds an attack time (fade-in) at the front of the envelope. When set to ZERO (fully anti-clockwise) it has no effect on the envelope contour—fully clockwise it gives up to two seconds of fade-in.

Envelope LED The brightness of this follows the envelope contour produced by the envelope follower. It is useful for setting the sensitivity control and visually displays the effects of changing the **Attack** and **Release** controls.

Release This adds a variable release "tail" to the envelope. Even after the control input signal has died away, up to five seconds of fade-out may be added by this control. Setting this control to small amounts can be useful for smoothing out rapidly changing envelopes when in ENV mode.

1. Env source
2. Sensitivity
3. Attack
4. Release
5. Env/Gate
6. Envelope LED

LFO (Low Frequency Oscillator)

Rate This changes the sweep rate of the low frequency oscillator, from about one cycle every ten seconds to 100 cycles per second. Note that when the **Link** switch is on, only the **Rate** control of the left (upper) channel has any effect as it is being used to control both channels.

Rate LED A bright LED indicates that the LFO is at the peak of the waveform (high pitch, if it controls the frequency, or high volume if it controls the VCA).

Waveform Selects between the four available low frequency oscillator waveforms—TRIANGLE, SQUARE, RAMP UP and RAMP DOWN. Note that when the **Link** switch is on, only the **Waveform** control of the left (upper) channel has any effect as it is being used to control both channels.

LFO Depth Determines the amount of LFO sweep sent to the filter and/or voltage controlled amplifier.

Link Normal/Invert *Only on the left channel!* Via the **Link In/Out** switch, the LFO of the left channel can be sent to the bottom channel (right). The **Link Normal/Invert** switch inverts the output of the left LFO, which is being sent to the right channel. This means that every time the left channel LFO is at its peak, the right channel's LFO is at its lowest phase.

Link In/Out *Only on the right channel!* When set to IN, The left channel LFO is sent to both channels (for stereo effects). This means that the **Rate** and **Waveform** settings of the left channel affect both channels. When set to OUT, the LFOs affect their respective channel. Please note that all other settings are still active for each channel, **Link In/Out** only links the two LFOs.

VCA/Both/VCF Selects whether the output of the LFO sweeps the voltage-contolled amplifier, the voltage-controlled filter, or both simultaneously.

Filter Section

Env Sweep Determines the amount of filter frequency sweep (up or down) being sent to the voltage controlled filter from the envelope follower.

Cut Off Frequency This sets the centre roll-off frequency of the voltage-controlled filter.

Resonance Sets the Resonance (or Q) of the VCF. High values produce self-oscillation.

VCA In/Out Selects whether the final output goes through the voltage-controlled amplifier or is taken from the VCF output.

1. Rate
2. Waveform
3. LFO depth
4. Link Normal/Invert
5. VCA/Both/VCF
6. Env sweep
7. Cut off frequency
8. Resonance
9. Rate LED
10. Link Out/In
11. VCA In/Out

Tempo Sync

Tempo Sync On/Off Setting **Tempo Sync** to ON means that the LFO rate of both channels are synchronized to the DAW host's tempo. With this set to ON, the **Rate** knobs of both channels are deactivated.

Tempo Sync Allows the user to set the speed of the LFO in sub divisions of the DAW host's tempo.

1. Tempo sync On/Off
2. Tempo sync
3. Width
4. Dry/Wet mix
5. Volume

Output Section

Width This is essentially a Mid/Side matrix. Set to 0 (noon), the channels are panned hard left and right, respectively. When turned counter clockwise, the channels are both panned gradually towards the center. Both channels reach the center when the knob is turned fully counter clockwise. Turning it clockwise increases the apparent spread of the channels, making them sound as if they are wider than the actual loudspeaker placement.

Dry/Wet Mix Turned fully clockwise, you will hear only the sound of the Mutator itself, which is how the original hardware worked. By turning the knob counter-clockwise, you can blend in the original, unaffected sound source to your liking.

Volume Sets the overall output volume of the Mutator plug-in.

Operation of the Mutator

Even though the Mutator has a lot of knobs and switches, it's really very easy to operate. Basic idea is this:

What do we want to change? Amplitude and/or cut-off frequency?

We can change the amplitude by enabling the *VCA* (voltage controlled amplifier), or change the cut-off frequency of the filter by enabling the *VCF* (voltage controlled filter).

How do we want to change it?

We can change it either by letting the "loudness" of the input signal determine the change, by using the *envelope follower*. Or we can let a low frequency waveform, the *LFO*, determine the change. Or both at the same time.

Envelope Follower

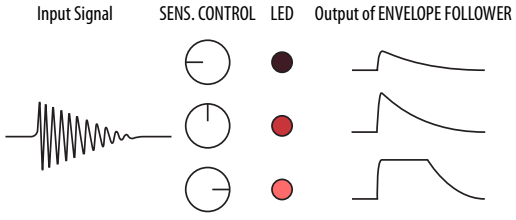
As previously mentioned, an envelope follower continuously tracks the amplitude of a signal and uses this envelope contour to control the cut off frequency of the filter. The signal that forms the base of this process is called the control signal. The control signal can either be the incoming signal itself (**Env Source** is set to INT) or an external signal that is fed to the Mutator using the sidechain functionality (**Env Source** is set to EXT). Extracting the countour from the internal signal could for example be used to create an auto-wah type of effect, where the louder parts of the sound source makes the cut off frequency of the filter move upwards.

A use case for the EXT setting would be to insert the Mutator on a piano track, and have the filter envelope on the piano be controlled by a drum loop on another track. In this case, the filtering on the piano would occur in time with the drum loop. Please refer to your DAW manual for instructions on how to set up a plug-in to accept an incoming sidechain signal.

Setting Sensitivity Control

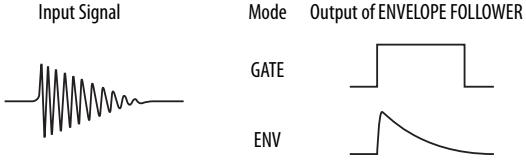
If the mode switch is in ENV position (and **Attack** and **Release** set to 0), turning up the **Sens** control will cause the LED to light when a signal is present—its brightness following the loudness of the signal.

For optimum performance, set the **Sens** control so that the LED reaches maximum brightness when the signal appears loudest. You may have to back the dial off slightly counter clockwise, as driving the circuit too hard will cause the LED to stay on maximum brightness even if the signal strength isn't quite at maximum.



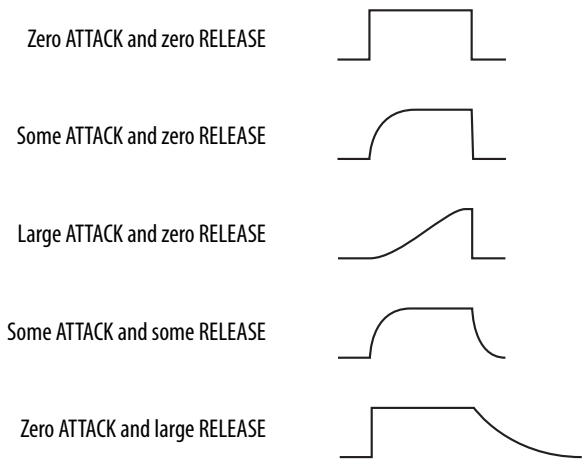
Gate Mode

If the mode switch is now flicked to the GATE position, the LED will come on at maximum brightness when a signal is present and be completely off when the signal falls low. Thus the output of the envelope follower will look like the diagrams below depending on the switch position.



Attack and Release in GATE Mode

In GATE mode, increasing the **Attack** and/or **Release** controls will change the shape of the output as shown below.


Zero ATTACK and zero RELEASE
Some ATTACK and zero RELEASE
Large ATTACK and zero RELEASE
Some ATTACK and some RELEASE
Zero ATTACK and large RELEASE

As you can see, a versatile range of envelope shapes may be constructed from the original gate shape.

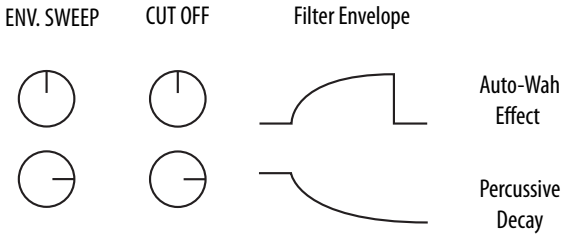
Attack and Release in ENV Mode

In ENV mode, the **Attack** and **Release** controls work as described above, but super-impose their contours on top of the original envelope shape of the signal. This can be very useful for smoothing out fast "wobblers" from a complex envelope!

Filter Sweep

The output of the envelope follower section is sent to its voltage controlled filter via the envelope sweep control in the filter section. This control has a centre zero (at noon) with both positive and negative sweep amounts available. Thus the arriving envelope contour can sweep the filter cut off frequency up or down from its original setting—which is determined by the **Cut Off** knob.

The same incoming envelope could produce entirely different effects on the sound depending on the settings of the **Env. Sweep** and **Cut Off** controls, for example:



If the **Env Sweep** control is set to 0, the filter cut off frequency will be manually changed by the **Cut Off** control.

The **Resonance** control allows a variable resonant peak to be added to the filter response, similar to that produced by an analogue synthesiser filter. At low settings the sound will appear fatter with subtle treble roll-off, much like a tone control. At higher settings a noticeable "twang" will be heard as the filter is swept, and at the highest settings the filter will begin to self-oscillate at its cut off frequency.

VCA In/Out Switch

After the filter stage, there is a voltage controlled amplifier (VCA) which, when switched in, controls the final output volume of that channel. The VCA is driven by the output of the envelope follower with the same envelope that is fed to the **Env Sweep** control. If an external source is selected for the envelope follower, and the gate mode is selected, and the VCA is switched in, the treated sound will be gated in and out by the external control signal.

The filter section can be effectively bypassed by turning **Cut Off** full up and its resonance to zero. Set this way, the VCF becomes "transparent" and with the VCA

switched in, the Mutator will function as a dedicated noise gate. If the cut off and resonance are altered, a filter sweep will combine with the gated effect.

The Low Frequency Oscillators (LFO)

The Mutator has two separate LFOs, one for each channel. The low frequency oscillators can automatically sweep the filter cut off (and/or VCA volume). Four waveforms are available: TRIANGLE, SQUARE, RAMP UP and RAMP DOWN. The sweep speed is controlled by the **Rate** control and the sweep range is adjusted by the **Depth** control.

The LFO output may be sent to the VCF cut off , the VCA level, or to both depending on the position of the **VCA/Both/VCF** switch.

Linking the LFOs

The LFOs of the two channels can be linked, in order to make them work in tandem. Note that the **Link** switches on the two channels are labeled differently and are used for different settings. The **Link In/Out** switch on the right channel (the second row of knobs) activates or deactivates linking. When activated, in the IN position, the LFO of the left channel affects both channels simultaneously. So with this setting, the LFO Rate and LFO Waveform of the right channel are deactivated. With this **Link** switch set to OUT, both channels' LFOs operate independently of each other.

The **Link Normal/Invert** switch on the left channel (upper row of knobs) can be switched between NORMAL and INVERTED. In NORMAL mode, the LFO affects both channels in exactly the same way. In INVERTED mode, the LFO's sweep is inverted for the right channel, essentially turning the waveform upside down. So every time the waveform is peaking on the left channel, it will be at its lowest level on the right channel, and vice versa.

Tempo Syncing the LFO

By turning on the **Tempo Sync On/Off** switch, the LFO rate controls are overrided by the DAW's tempo information and the Rate is no longer doing anything. By switching the **Tempo Sync** knob you can change the full period of one waveform to become a 1/16th note, 1/8th note, quarter notes, and all the way up to 4 bars. For extreme settings (say 4 bars at 30 BPM) the waveforms of the LFO might change a bit.

Resetting Waveform

By hitting play, most DAW's will send a reset command to the plug-in which will make the LFO start from scratch. For example, if the **Tempo Sync** is set at 1 BAR, you will get a different result if you start playback on the downbeat or in the middle of the bar.

Output Section

The output section consists of a **Volume** control, a **Stereo Width** and a **Dry/Wet Mix**. If the left and right channels coming from the Mutator are very different, and you want to reduce the stereo width, the natural way would be to set the **Stereo Width** to somewhere between MONO and 0, but in many cases you can achieve a similar (but not exactly the same) effect by settings the **Stereo Width** more to the WIDE setting. The main difference between MONO and WIDE is that in both channels will output L+R in MONO, while in WIDE one L channel will output L-R, while the other channel outputs the negated signal R-L.

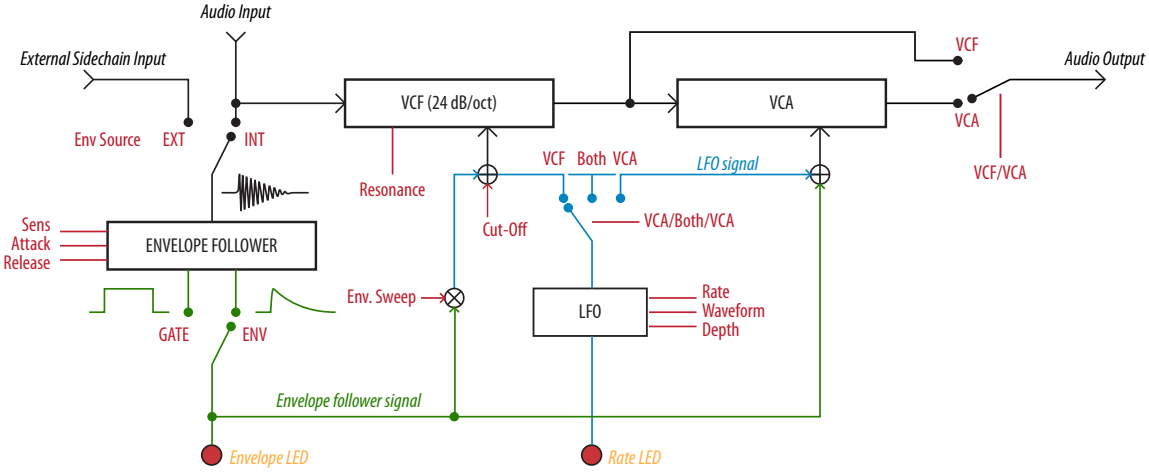
Block Diagram

The block diagram is a simplification of how it really works, but a useful help to figure out which knob to adjust.

One thing to note is that the Env Sweep control only affects the envelope follower, not the LFO.

Credits

Arvid Rosén – modeling. **Henrik Andersson Vogel** – manual, project management. **Niklas Odelholm** – graphics design, presets, math stuff. **Paul Shyrinskykh** – quality assurance. **Oscar Öberg** – math stuff. **Russ Hughes**, **Erik Putrycz** and **Paul Russell** – preset design. **Ulf Ekelöf** – graphics rendering. **Torsten Gatu** – framework programming. **Patrik Holmström** – framework programming. **Mattias Danielsson** – technical support.



39 OTO Biscuit 8-bit Effects

OTO BISCUIT CAME OUT IN 2010 and was the first product by French audio wizards **Denis Cazajeux** and **Stéphane Briat**. Denis had started out building stompboxes and synths for his brand *Cazatronics* already in the mid-2000s. One of the many boxes he built was the Biscuit, an effect housed in a plastic butter dish.

The original idea for the Biscuit was to simulate the sound of an old **Fairlight CMI** sampler. An 8-bit analog to digital converter was set up with a parallel output to an 8-bit digital to analog converter, and the sampling frequency could be set with a dedicated knob. A 12 dB per octave low-pass filter with a Q control was added to the prototype, as well as switches for muting and

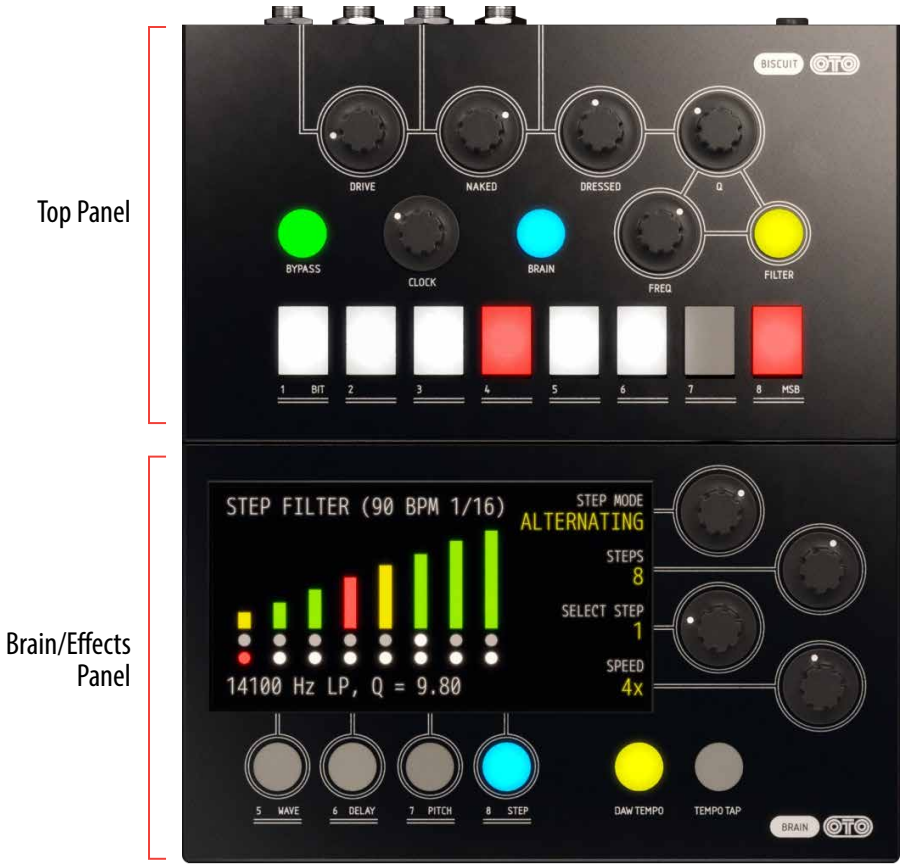
inverting the bits in the eight lines between the AD and DA converters.

But it wasn't until several years later, when Denis teamed up with producer/engineer Stéphane in a studio near Paris, that the two started expanding the Biscuit concept and turning it into a commercial product.

Overview

OTO Biscuit is a useful tool for all sorts of noise experiments. It's particularly useful with electronic sounds, but don't let that stop you from trying it on any sound source you can think of. At its core, the Biscuit is a bitcrusher, but some clever additions extends its use far beyond that.

It produces distortion by reducing the resolution and bandwidth of digital audio data. The resulting quantization noise can be varied from adding the familiar warm 8-bit character and lo-fi digital artifacts, to harsh numeric distortion. The "bitcrusher" section of the Biscuit is followed by an analog multimode filter (with low pass, high pass and band pass modes) and a simple mixing section (dry/wet). Furthermore, each of the eight data bits can be removed or inverted for more variation to the character, and there are four on-board digital effects—waveshaper, delay, pitch shifter and step filter.



1. Bypass
2. Drive
3. Clock
4. Naked
5. Brain
6. Dressed
7. Freq
8. Q
9. Filter
10. LSB (least significant bit)
11. Biscuiting switches
12. MSB (most significant bit)

Top Panel Controls

Drive Input gain knob. With a normalized signal gained above a certain level, the emulated diode clipping will be noticeable in the audio-chain and the **Bypass** button will flash red.

Naked Sets the amount of the dry signal mix, from silent up to 0 dB.

Dressed The amount wet signal mix, from silent up to 0 dB.

Q Resonance of the multi mode filter

Bypass This button bypasses **Biscuit when unlit**, green light means that Biscuit is active and processing. The button flashes red to indicate that the emulate drive circuit is clipping. It is usually desired to have some clipping happening.

Clock Sets the internal sample rate from 250 Hz up to 30 kHz. Changing the internal sample rate introduces additional gritty sonic artifacts as side-effects. This is a very recognizable effect that can be heard on many records.

Brain Toggles the Effects section on and off and thereby engages the last selected effect when lit.

Freq Cutoff frequency of the multi mode filter

Filter Sets the filter type. Its color indicates which type is selected:

GREEN: low pass

YELLOW: band pass

RED: high pass.

Biscuiting Switches

1-8 These switches mute or invert the 8 bits of audio stream through the Biscuit.

WHITE (lit) means that the **Bit** is ON

RED means that the **Bit** is INVERTED

GREY (unlit) indicates that the **Bit** is MUTED.

Inverting or muting bits of data can result in very harsh and loud noise, please be careful.

Inverting or muting the MSB (most significant bit), the bit furthest to the right, will result in more dramatic changes in sound, while doing the same to the LSB (least significant bit) yields more subtle results.

SHIFT-clicking on a switch will change all switches to the same value at once.

Brain/Effects Panel

Biscuit contains four different effects, of which one at a time can be used with the sample rate reduction, bit-crushing and analog filter:

1. WAVE SHAPING
2. DELAY
3. PITCH SHIFTING
4. 8 STEP SEQUENCER FOR THE FILTER

Use the wave shapers to transform your sound, from crazy distortion to sub-octave effects. Add a gentle lo-fi delay or a screaming 100% feedback delay. Pitch shift your melodies to add chorus or harmonies, or use the step sequencer to add DAW synchronized filter movement on your sounds.

The **Brain** button engages the last used effect when lit, and turns it off when unlit. The effect section can also be engaged by clicking on the desired effect button below the EFFECT DISPLAY.

Wave Shaper

Delay

Pitch Shifter

Step Filter

1. Effects display
2. Effect dependent parameters
3. Wave
4. Delay
5. Pitch
6. Step
7. DAW tempo
8. Tempo tap

Effect Display	Shows the user information on the currently active effect and its related parameters that are linked to the EFFECT KNOBS to the right of the display.
DAW Tempo	Syncs the tempo to the DAW tempo for the DELAY and STEP FILTER effects.
Tap Tempo	Tapping this button sets the tempo for the DELAY and STEP FILTER when **DAW Tempo** button is disabled.
Wave	Turn on the Waveshaping mode
Delay	Turn on the Delay mode
Pitch	Turn on the Pitch Shifter
Step	Turn on the Step Filter mode.

Each mode has one to four "soft parameters" that are specific for the different modes. For a description of those parameters, see the section corresponding to the effect mode.

Wave

The waveshaper effect can be used to distort, alter or dynamically track and filter your signal. Engage it by pressing **Wave**.

Wave Type	Select the 8 different wave shapers. Shapers 1 to 5 will create distortion, shapers 6 to 8 synthesize a wave form.

1. RECTIFYING: the negative parts of the incoming signal is inverted and added to the positive part. This creates an octave up effect similar to a fuzz octave pedal.
2. ALTERNATE RECTIFYING: only positive parts of the incoming signal is used. Creates a distorted "octave down" effect.
3. BAT FUZZ: the input above a threshold gets folded.
4. BISCUIT: "Fifth down" effect with some additional crunchy distortion.
5. SWAP: Bits 1 to 4 are swapped with bits 5 to 8 creating some radical distortion with a lot of high harmonics.

Waveshapers 6 to 8 in the Biscuit are small synthesizers, creating waveforms that track to the incoming signal frequency. These three waveshapers work best on bass synth sounds but can lead to exciting new sounds on other sources.

To allow proper tracking, crank up the drive knob properly so that the bypass button shows red to obtain a stable sound.

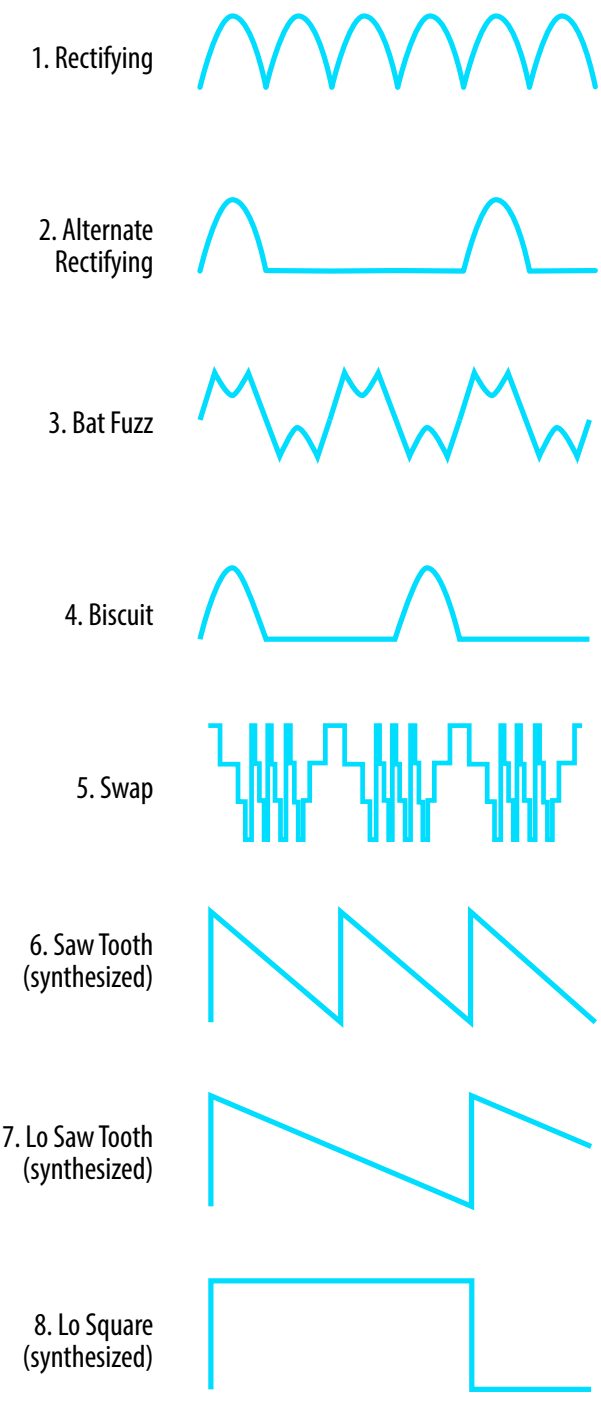
6. SAWTOOTH: Tracking SAWTOOTH waveform.

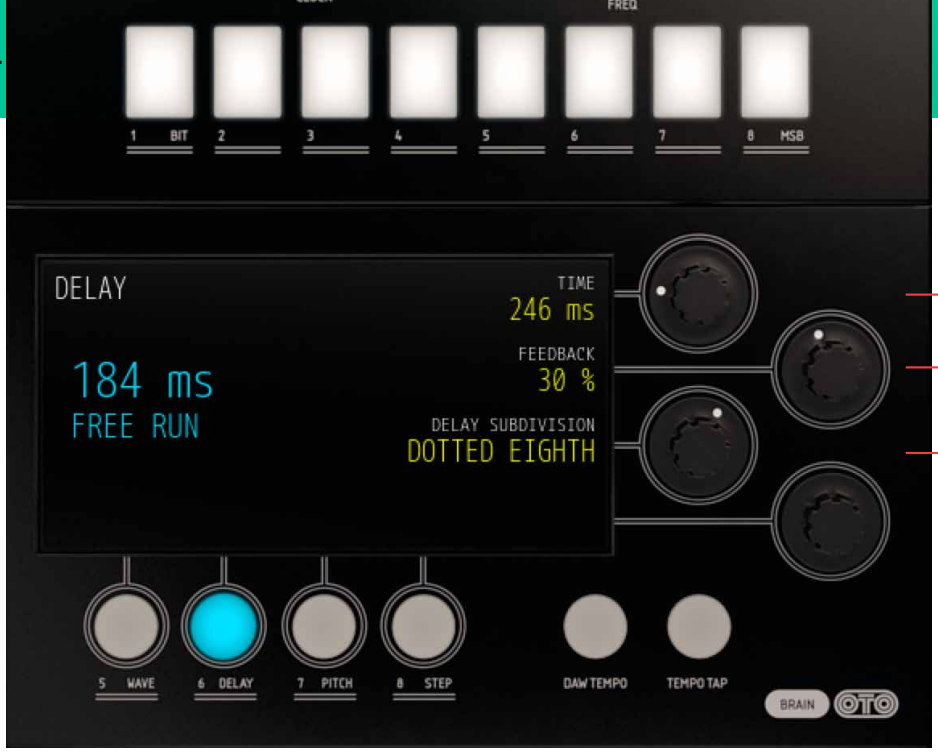
7. LO SAWTOOTH: Tracking SAWTOOTH waveform one octave down below source.
8. LO SQUARE: Tracking SQUARE waveform one octave down.

VCF/ENF Sets the amount of the envelop output that will affect the filter cutoff frequency. This effect is most apparent when lowering the cutoff frequency (**Freq** knob) on upper panel.

The **VCF/ENF** knob is only active in modes 6 to 8.

When using both the **Biscuiting** switches (bitcrusher) and **Wave Shapers** at the same time, it might be difficult to understand how these two effects interact. In order to have a better understanding of the **Wave Shapers**, we recommend to start using them with all the 8 bits in their normal state (all the 8 switches lit up white) at first.





1. Time
2. Feedback
3. Delay subdivision

Delay

The delay effect in Biscuit is a 8-bit stereo delay as opposed to the original hardware mono operation.

Only the delay will be heard via the **Dressed** Knob. This means that, if you want to hear both the source signal and the delayed echo at the same time, turn up the **Naked** and **Dressed** volume knobs accordingly.

> Sample rate reduction and filtering will only apply to the delayed signal in **Delay** mode.

Time Sets the delay time of a quarter-note in free running mode. It also controls the time of the STEP FILTER (see below) free running mode which is using the same internal clock.

Time knob is disabled in DAW SYNC mode (as delay time is set automatically when **DAW Tempo** is lit).

Feedback Sets the amount of feedback, from 0% (one echo) to 100% (infinite build).

Delay Subdivision Sets the subdivision of the delay time into:

QUARTER NOTES (preset),
DOTTED QUARTER NOTES,
EIGHTH NOTES,
DOTTED EIGHTH NOTES,
SIXTEENTH NOTES, and
DOTTED SIXTEENTH NOTES.

The actual delay time is show to the left in the effects display.

Pitch

The PITCH SHIFTER is a 8-bit mono effect that transposes the audio signal by using a short delay buffer and sends it to both left and right outputs via the **Dressed** knob.

Shift Sets the amount of pitch shift through a number of presets, ranging from -2 octaves up to +1 octave.

You can either use the knob, or click directly on the dots on the displayed keyboard to change pitch. Middle C is for creating chorus-like effects by mixing the **Naked** and **Dressed** signals. Click twice on the lowest C to alternate between -1 octave and -2 octaves pitch shift.

In the original hardware the sample rate in the pitch shifter is fixed at maximum rate to obtain as good pitch shift effect as possible.

Using both Biscuiting and sample rate reduction will make the pitch shifting algorithm work more like a pitch shifting delay, thus creating some cool effects not obtainable in the original hardware unit.

Step

The STEP FILTER is an (up to) eight steps filter sequencer where each step can be set to have its own **Filter Type**, cut off **Frequency** and **Q** value.

In STEP FILTER mode the effect display will show the BPM of the sequence related to division and/or DAW sync (if selected).

Each step in the sequence is shown in the display represented by a bar and two dots.

- The bar height represents the **Frequency** of the filter.
- The bar color represents the **Filter Type** (green for LP, yellow for BP and red for HP)
- The bar light intensity represents the **Q** value of the filter.
- The top row of dots indicates the "song position", ie. which step in the sequence that is currently playing.
- The bottom row of dots shows the currently selected step, the step that can be edited by adjusting the filter knobs. It also shows the total length of the sequence.

You can click directly on the display to select current step and change filter **Frequency**. To set **Filter Type** and **Q** value per step, you need to use the knobs on the upper panel.

A sequence can be 2, 3, 4, 6 or 8 steps long.

Step Mode Sets any of the four playback modes:

1. FREEZE: in this mode only the selected step is played. This is convenient for setting up the step's parameters (**Frequency**, **Q**, **Filter Type**) in detail.
2. FORWARD: the sequence is played from left to right.
3. ALTERNATING: "pendulum", sequence alternates between playing forward and backwards.
4. RANDOM: deterministic random, steps are played in random order.

Steps The amount of active steps of the sequence: 2, 3, 4, 6 or 8 steps long. Active steps are lit up and marked by white dots in the lower row beneath the display. Steps not active in the sequence are dimmed grey in the display.

Select Step Select the step to be edited. This can also be achieved by clicking in the display or on one of the rows of dots below the display.

Speed The speed division for each step in the filter step sequencer. Default is that each step is a quarter note which corresponds well to BPM free running.

When DAW sync is used division information (quarter, eighths, sixteenth and so on) is represented in the upper right corner in the effect display along with synced BPM.

Setting the Tempo in Free Running Mode

When the **DAW Tempo** is inactive (unlit), the STEP FILTER is in a free running mode using the same internal clock as the free running delay. This means that you can set the tempo either by tapping the **Tempo TAP** button or by adjusting the delay **Time** knob in the DELAY effect.

For an exact setting of tempo in the step filter, DAW sync is recommended.

1. Freq
2. Q
3. Filter
4. Step mode
5. Steps
6. Select step
7. Speed
8. Frequency
9. Select step
10. DAW tempo
11. Tempo tap

Linking Parameters

Some knobs and buttons have extra functionality (parameter linking) that can be accessed by shift clicking the knob/button.

Clock knob: change the clock speed and filter cutoff simultaneously.

Naked or **Dress** knobs: change both knobs at once with an inverse relationship to mirror that of a cross-fade operation.

Biscuit switches (bit inverter/mute switch): change all Biscuit switches to the same value at once.

Freq, **Q** or **Filter** in STEP MODE: set all steps to the same value. This can be useful for example when you want to go from a complex sequence with a lot of different use filter modes to a simpler one.

Tips

Mix the **Dressed** and **Naked** signals to achieve phased wave type sounds when sweeping cutoff and/or clock rate.

Inverting and removing bits will cause more noticeable effects the more significant the bits are (that is, the further right the buttons are).

When turning off or inverting the **MSB**, lowering the **Drive** level can act like a kind of a way to "gate" the signal and thereby dramatically alter the dynamics of the sound. Try it, you might like it!

The step sequencer can also be used for "gating" sounds, such as pad, in sync with the DAW. Use LP mode and set **Freq** on the desired "silent" steps to 0.

Using pitch shift and clock rate reduction at the same time can create some interesting results not possible in hardware.

By using clock rate reduction and HIGH PASS filtering of drums, nice additional hi-hat sounding parts can be created.

Using a combination of the SWAP wave shaper and the HIGH PASS filter mode can create nice additional high end "shakers" when applied to drum loops.

Credits

Björn Rödseth – modeling. **Kim Larsson** – modeling. **Arvid Rosén** – modeling. **Kristofer Ulfves** – product owner, user manual, testing and presets. **Niklas Odelholm** – graphic design. **Paul Shyrinskykh** – quality assurance. **Ulf Ekelöf** – 3D rendering.

40 Parallels

Parallels is all about sonic exploration. Softube Parallels is a plugin synthesizer featuring high quality pre-recorded multi-waveforms in the Source-section, custom made filters in the Shaper-sections and a special selection of different effects in the, you guessed it - Effects section (right hand side panel). Most of the parameters in the different sections can be modulated by one of the five modulator types (LFO, RND, EUC, ENV, SEQ) available in the Mod Pod section (left hand side panel).

We hope you will find Parallels inspirational and that you will have fun sound exploring its hidden depths!

Foreword by Johan Antoni, concept creator of Parallels

I have been using, selling and buying synthesizers as long as I can remember. My father had a music store and imported the famed EMS Synthi-synthesizers (among others) in the mid 70s. You could say that I grew up with synthesizers and that they occupied much of my time since I was a kid. Since 1996, I followed in my fathers footsteps and have been running (dare I say?) Stockholms premier synth shop, selling mainly vintage, quirky and boutique synths. Or any synth I find interesting, really. Lately, its been mostly expensive or rare modular systems.

A couple of years ago, after having played around with a Korg 800DV and an Oberheim TwoVoice in the store for a couple of lazy summer days with no customers, I realized that I sort of missed a less complicated and instead more direct way of doing more complex sounds. As you may or may not know, both of these synths mentioned are old 70s vintage analog synths with a kind easy-to-follow architecture, very versatile filters (even compared to todays standards), not that many parameters and they work by giving you two of everything. In short, they are very competent, user friendly, and they allow you to layer two sounds on top of each other in order to make complex sounding sounds very quickly and easy.

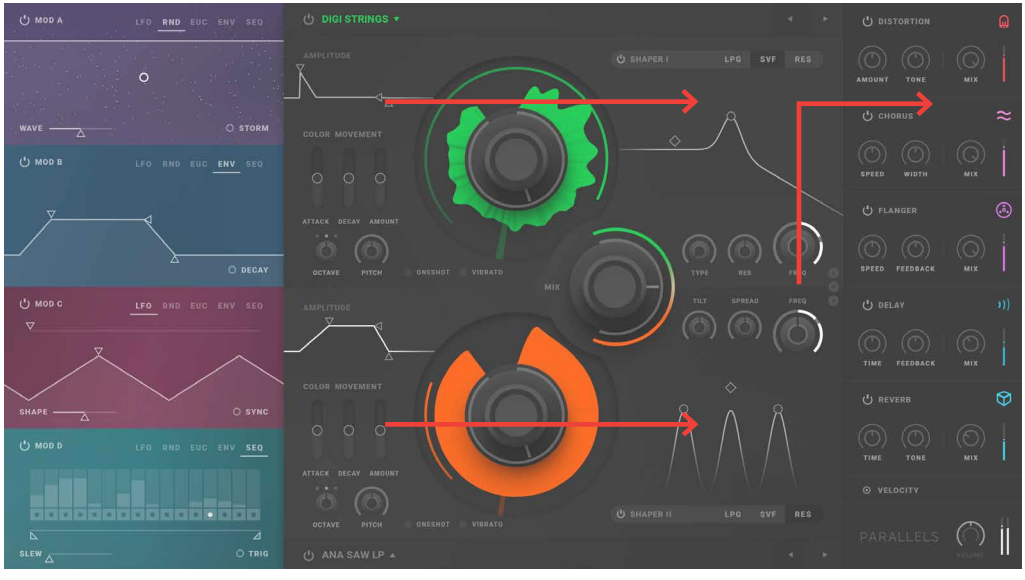
Combined with my idea of the ”source waveform” based synthesis (explained later in this manual) which allows you to switch quickly between different snapshots of synthesis, this became a concept I could not let go of. I wanted this as a software plug-in. After some very basic software idea tests I approached my friends at Softube and they loved the concept; why not start developing this right now?

For my part, what followed was producing the ”source waveforms”. This took place mostly after hours in the store using whatever I could think of (mostly vintage synth gear), some basic programming at home and a period visiting friends in Tokyo.

User interface

1. Mod pod
2. Source section
3. Shaper section
4. Effect section

Parallels signal flow chart



Introduction to Parallels's different Sections

Parallels is a very simple, but at the same time, complex 14 voice polyphonic synth. Polyphony is shared among two layers, upper and lower. Parallels's simplicity lies in that it has a very logical signal flow for the audio. However, for the complex part, Parallels also has a modulation resource, the Mod Pod, with multiple slots that can be customized and routed to a huge number of available destinations. Here follows a brief description of the different sections of Parallels:

Source section

Let's start looking at the Source section: placed in the middle left at the top and bottom, Parallels's Source section and is the heart of the instrument. Each Source section, the upper and lower parts, contains a waveform that is selected through the Wave dropdown menu or via the wave selection arrows. This waveform is displayed as wrapped around so called the Color knob which clearly indicates the waveform start-point and modulated loop. Each Source section also have a dedicated Color modulation envelope and a dedicated amp envelope.

Shaper

The Shaper section contains three different filtering modes – **Low Pass Gate (LPG)**, **State Variable Filter (SVF)** and **Resonator (RES)** - each one with three modulatable parameters.

Mod Pod

This Modulation Pod (alias Mod Pod) contains five different kinds of modulation that can be routed virtually everywhere in Parallels (with a few exceptions). It contains four Modulation Slots named Mod A to D which can contain any of the five different modulation types at ones. The Modulation types are a Low Frequency Oscillator (LFO), a digital Random Source (RND), a Euclidian sequencer (EUC), a polyphonic modulation envelope (ENV) and a modulation sequencer (SEQ).

Effects

The effects section contains five different effects in series that can be turned on or off/bypassed, each with control over two parameters and dry/wet portion. The effects are Distortion, Chorus, Flanger, Delay (unsynced/synced) and Reverb.

MIDI

Parallels is a virtual instrument that interfaces with your DAW through the MIDI protocol. Many DAWs load these types of "virtual instruments" on to so called Instrument tracks which often are set up to send MIDI information by default to the plugin instrument on channel 1. Parallels actually receives its MIDI information on all 16 channels (MIDI omni mode).

The Velocity option turns on and off Velocity to amplitude response. This mean that, when Velocity is set to on, soft strokes on your MIDI keyboard will create pretty quiet notes while harder strokes will create louder notes. When Velocity is turned off, any change in velocity will thus not be reflected in quiet and loud notes.

The keyboard range of Parallels covers a little more than five octaves (G0 – C#6). MIDI notes received outside this range will not be registered and thus not heard.

The pitch bend range is always + - 2 semitones.

Modulation wheel (MIDI CC#1) is always routed to vibrato amount, engaged when vibrato is turned on. The present value of modulation wheel is not saved in the patch.

Interface

Press CTRL/CMD to have fine-adjust mode activated (as in our other plugins) which also will prevent more than one step being changed at the same time in the sequencer.

Furthermore, for controls that change two parameter at the same time (such as for the shaper controls), it can be "locked" into just X or Y direction by pressing SHIFT when dragging. These two controls can also be combined (where applicable).

Getting started

Basics – what is this Source Waveforms, Shapers, Envelopes, Mod Pods etc???

Sound is a vibration that typically propagates as an audible wave of pressure, through a transmission medium such as a gas, liquid or solid. In digital audio (as in Parallels), sound is signal that has been recorded as or converted into digital form. This means that the sound wave of the audio signal is encoded as numerical samples in continuous sequence stored in a digital medium (for example a computer hard drive).

Parallels's Source section uses pre-recorded digitized audio that has been meticulously recorded in high fidelity and prepared to cover multiple octaves in order to ensure high quality and great musical performance. Each waveform has been selected to be musically inspiring to work with and also each represent a sonic transformation in the sonic spectral properties, but also sometimes in amplitude and pitch. Each waveform can use any startpoint and be swept back and forth freely by the dedicated Color envelope shape or any modulation.

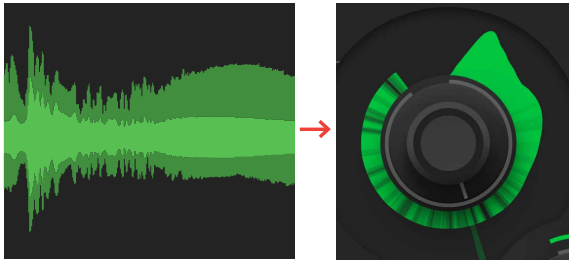
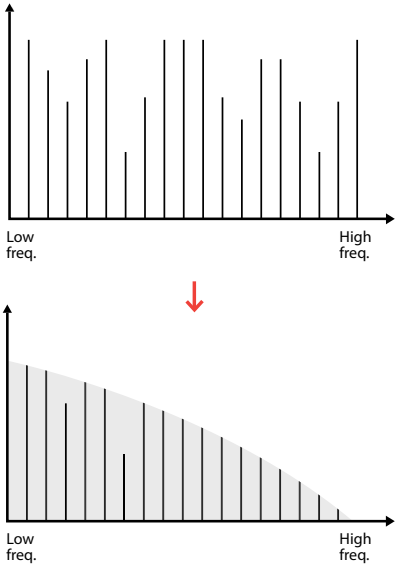


Figure wave to waveshape.

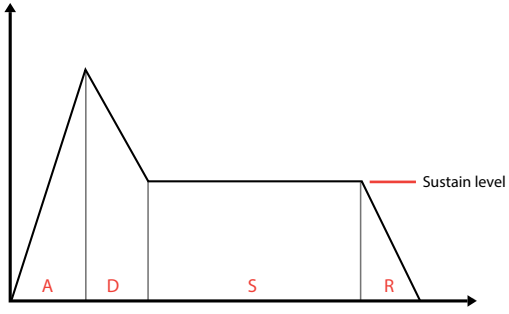
The Source waveforms in Parallels creates a sonically rich spectrum. This spectrum can be further shaped into containing the desired amount of overtones and amplitudes with the so called Shaper section. The Shaper section contains different kinds of filters (and distortion). Each filter affects the spectral outcome of the Source waveforms in different musical ways.



Example of lowpass filtering.

An envelope generator describes a triggered series of timed events, in instruments normally triggered by the note start. The envelope is used to represent the natural sound properties in a formalised model. There are different models and school of thougths when it comes to envelopes in synthesizers. Parallels uses several different ones (AD, ASR and ADSR) for flexibility.

The **Attack** portion is the rise time of the envelope, i e the time it takes for it to reach its peak. The **Decay** time is the time it takes for the envelope to fall from the peak down to the next level, often to be the **Sustain** level. The Sustain level is the level at which the note is statically suspended and sustained until the note is released (key is released). The last envelope phase is the **Release** phase, describing the time it takes from the note is released until it has all died out.



ADSR Envelope.

Getting Started: Let's get to know Parallels's Source section!

1. We will start off by turning off all modulation and all effects.

2. Continue on by turning off the lower Source section and the top Shaper. Now when you play Parallels, all you should hear is the raw waveform of the upper Source section.

3. Turn down the upper Source Movement Amount slider to zero. Now try to move the big Color knob clockwise and counterclockwise back and forth while playing. You will notice that the sound goes from a duller string sound to brighter one the further clockwise you will go. And at the very end the waveform is silent.

Now turn the Color knob back all the way counter-clockwise (Color 0%).

Now image that you could do the clockwise and counterclockwise movement back and forth motion you previously did with your mouse in an automated manner - Yes, this is precisely what the Color Movement Envelope is made for! The Movement Attack time is the timing of the clockwise movement, the Movement Decay time the timing of the counterclockwise movement and Movement Amount the striking distance (i e the multiplier) of the first two.

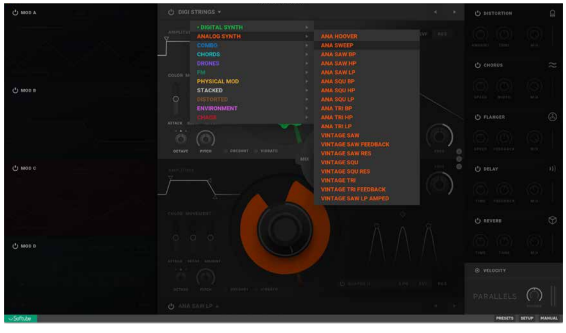
4. Try out the Color Movement Envelope by setting a fairly long Movement Attack (~1300ms) and Decay (~2500ms), with a fairly high Movement Amount (75%). Now play a long sustained note, listen and watch as the playhead of the Source waveforms follows your instructions and now sweeps forward clockwise at first, then turning counterclockwise and returning to the start-point set by the big Color knob.

5. Now, try out different settings where you experiment with different Color knob settings for different start-point of the sweep and different Movement Amount for different strike range. Also try different time ratios of the Movement Attack and Decay. You'll notice that when Movement Amount is too high in combination with a high value on the Color knob (initial startpoint) the Movement envelope can sometimes sweep outside the waveforms boundaries resulting in a silenced or partially note. This can however be used to our advantage (more of this later).

6. OK, let's try something else. Set the Movement amount to zero again and the Color knob at default position 50% (alt-click on the knob to set it back to initial value).

7. Now, adjust the amplitude-envelope at the upper left corner of the Source section, into a smoother curve. This is done by adjusting the attack time and sustain level to match the picture above. Play a note. You will notice that the note now is quieter in the beginning, gradually fading up to sustained level and staying there until the note is released.

8. Now, adjust the amplitude-envelope curve to match the picture above again: Attack time at zero, sustain level at 100% and release time at zero. Play a note again and you will notice the instant full level amplitude response of the waveform now. And when you release the note, the sound is silenced at once.



9. Now, using the techniques you learned above, try out and explore the other waveforms by browsing the different waveforms in the waveform menu or using the arrow buttons in the upper right corner of the Source section. You will notice the vast amount variation available already within Parallels's Source section itself, and we've only just begun!

Getting Started: Now let's turn our attention to Parallels's Shaper section.

1. Let's turn on the Shaper section again. SVF mode is the default Shaper mode for the upper Shaper section.

2. Now try clicking on the hotspot in the displayed SVF (state variable filter) curve - it's on the crest or peak of the curve. Try clicking and moving it around while playing. You'll notice that by changing this graph in horizontal direction changes the SVF cutoff frequency while changing the graph in vertical direction changes the SVF resonance. Note also that the cutoff frequency and resonance knobs changes with the curve and vise versa.

3. Click on the other hotspot in the filter diagram and try to move it. You'll notice that it is linked to the Type knob and locked to be modified only horizontally. When the Type parameter is changed from a low value to a high value, the SVF topology is seamlessly changed from a lowpass filter, on to a bandpass filter and finally a high pass filter. The displayed graph also reflects this behaviour.

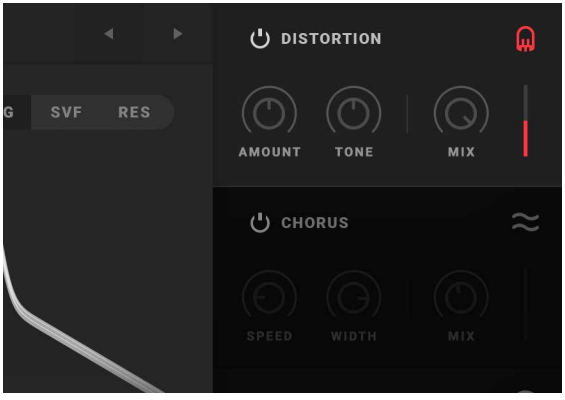
4. OK, Let's look briefly at the Shaper RES mode, the Resonator. Click on the RES tab on the Shaper mode selector on the top right, an interactive three peak filter curve appears. The Resonator shaper mode is great for creating formants or vocal sounds, but can also be used for crude phasing duties. By clicking on the middle peak hotspot and dragging your mouse or touchpad left and right, the RES Center frequency is changed. This is also reflected on the Freq knob in the lower right part of the Shaper window.

5. Let's click on the hotspots on one of the side-peaks in the filter curve. Clicking and dragging your mouse or touchpad reveals that moving the side-peaks hotspots vertically changes the RES Tilt (peak emphasis), while moving them horizontally changes the RES spread (the space in frequency between the different peaks). This change is, of course, also reflected on the adjacent knobs on the lower middle and left in the Shaper section.

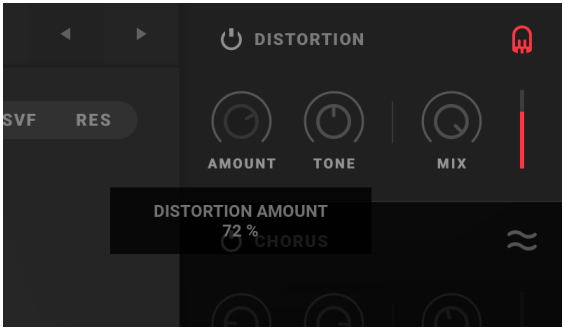
6. At last, let's look at the Shaper LPG mode. LPG stands for Low Pass Gate which is a special vactrol-controlled filter and VCA combination controlled via a so called vactrol, often used in so called West-Coast modular synthesis. Parallels's Shaper LPG mode draws some inspiration from the classic LPG to create a filter with a controlled "sloppy" cutoff response. Click on the LPG tab on the Shape mode selector on the top right to bring up the LPG filter curve.

7. Clicking on the different hotspots in the filter curve for the LPG reveals similar behaviour as the other two modes. The peak hotspot moved vertically controls the LPG resonance, while moving it horizontally, it will control the LPG cutoff frequency. The other hotspot on the left is locked to only vertical movement as it reflects the LPG Slew, i e the amount of "sloppyness" behaviour the Shaper LPG will display when its cutoff frequency is modulated (more of this later). Let's move on to the Effects section!

Getting Started: Getting to know Parallels's Effects section!



1. OK, let's start by clicking on the activate-button on the top effect in the Effects section (on the right in Parallels), distortion. When the color graphics light up (in the case of distortion, it is red), the effect is active.

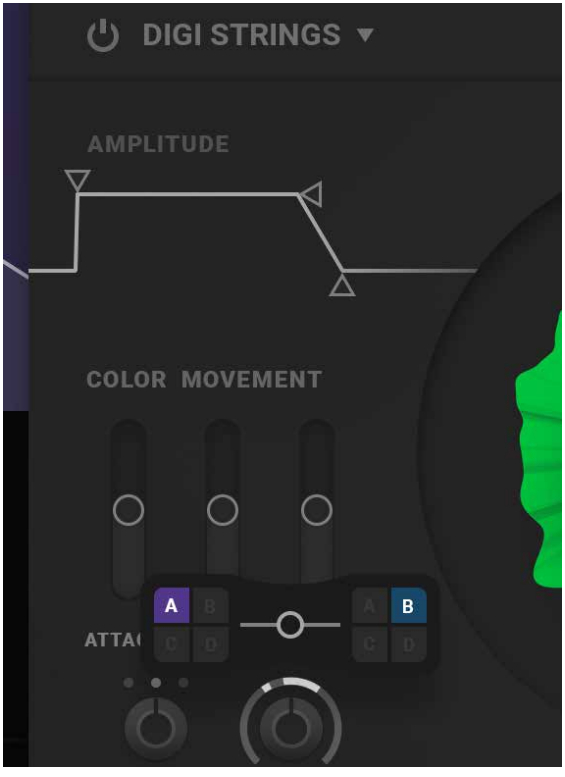


2. Each effect features two specially selected editable parameters and a dry/wet mix (the knob on the right). All the effects in Parallels is chained in series with a signal flow from top to bottom, which is good to know when working with effect.

Getting Started: Looking at Parallels's Modulation section

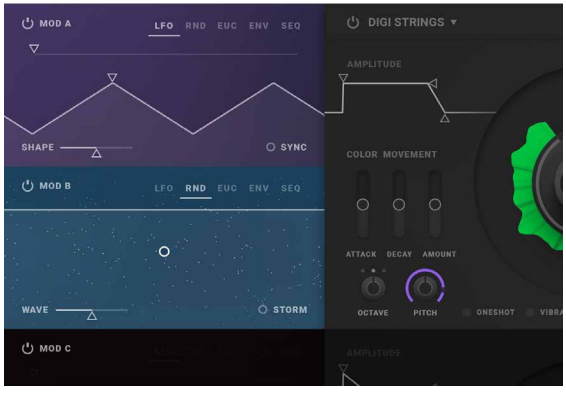
On the left panel we will find the Modulation Pod (or Mod Pod for short), containing all the four different modulation slots named A, B, C and D. Most of the parameters in the different sections can be modulated by one of the five modulator types (LFO, RND, EUC, ENV, SEQ) available in each slot of this Mod Pod section.

1. Let's get started by adding an LFO in Mod Pod slot A (the slot on top).

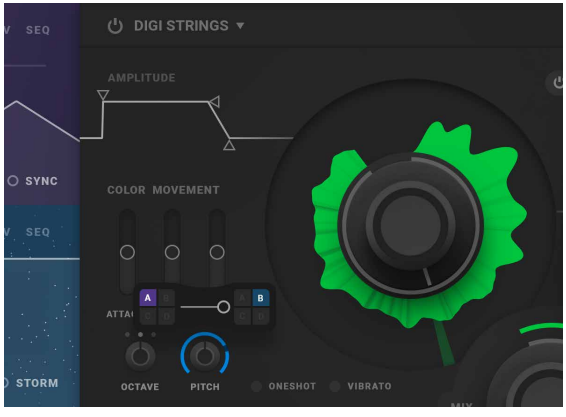


2. Now, let's add some modulation to the pitch of the upper Source. This is done by clicking on the outer ring of the pitch and dragging upwards. You will see the modulation ring start to fill with white color from the middle and out.

3. By adjusting the Modulation mix slider between the Modulation sources you will see that the color of the modulation will change accordingly. When the Modulation mix is turned all the way towards left where Mod Pod Slot A is selected as modulation source, the modulation color turns violet; the same color as the Mod Pod Slot A has. Play a note and you will hear the LFO affecting the pitch up and down, like a siren. Experiment by changing the speed, shape and rise/fall times in the LFO while playing.



4. Now, let's add another modulation source. Click on the Mod B Active button to active Mod Pod Slot B and select RND as Mod B Type as pictured.



5. Now, click on the Pitch knob outer ring again to bring up the modulation sources and mix controls for this destination. Slide the Modulation mix all the way to the right, towards mod B as modulation source, and you will see the modulation color turn blue; the same color as modulation slot B. Now, when playing a note you will hear a highly buzzing erratic sound due to the noise-source affecting the Source waveform pitch. Experiment by changing the different parameters in the RND source now selected in modulation slot B.

6. As we noticed before, when the modulation mix slider is set somewhere in the middle, in between the modulation source selected on the left and on the right, the modulation color is white; indication that there is a mixture of two modulation sources going on here.

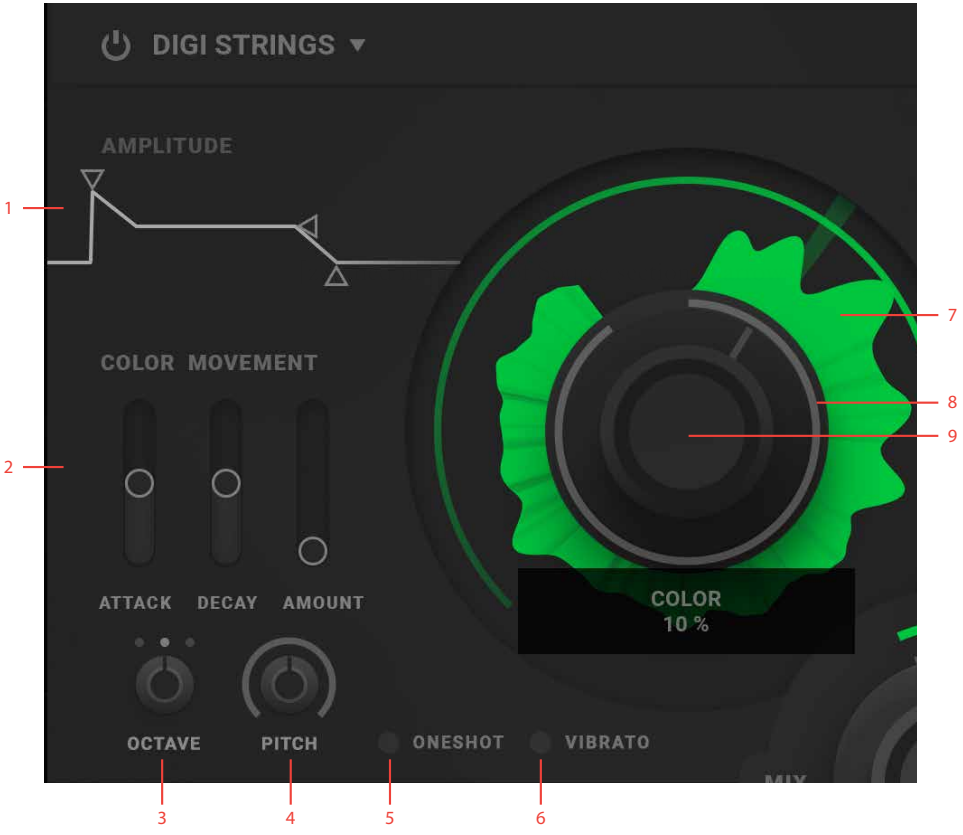
7. As pictured and described, you can route any modulator to any destination by clicking on the other ring of a parameter of choice. Then chose the modulation from two of the four modulation slots and the mix balance between them. Set the amount of modulation at the destination by clicking on the other ring of parameter again, and drag your mouse or mouse-pad upwards (the ring around the parameter will fill up).

Note that some parameters cannot be modulated - all non-knob-type interface will not accept modulation routing from the Mod Pod. This is true with the exception of the Octave knob that's not available for modulation (despite being a knob).

Parallels in detail

In these sections that follows, you'll find all detailed information about each and every part of Parallels.

The Source Section



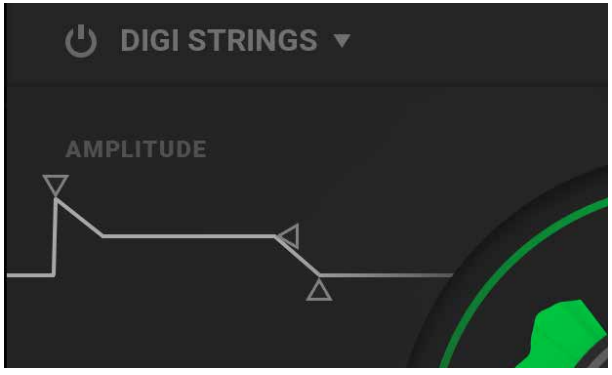
1. Amplitude envelope
2. Color movement envelope
3. Octave
4. Pitch
5. Oneshot
6. Vibrato
7. Waveform
8. Modulation
9. Color knob

Active on/off	This parameter activates and deactivates the Source with its adjacent Shaper section. When a section is deactivated, its graphics is dimmed indicating that it is no longer heard.
Wave menu	In this dropdown menu you can select from each of the different wave category.
Wave selection arrows	With the wave selection arrows you can rapidly change between waveforms without using the drop-down menu. Notice that a changed waveform will not be heard until a new note is pressed.

A complete Waveforms list is presented in "Appendix" on page 459.

The Amplitude envelope

The amplitude envelope controls the overall change of amplitude over time. It is a so called ADSR (Attack, Decay, Sustain, Release) envelope with simplified controls (i e one common control for Decay and Release times).



Amplitude envelope attack (1ms - 16sec) This horizontal slider controls the attack portion of the (upper) layer's amplitude envelope. Keep this short (low values) for instantly responsive and snappy sounds, while using higher values for extending the attack time for slower, fading in notes.

Amplitude envelope sustain (0-100%) This vertical slider controls the sustain level of the amplitude envelope for the upper layer. This is the sustained volume level that is held after the initial attack and decay times of the envelope. If the sustain level is at 100% the decay will not be pronounced, effectively turning the envelope into a ASR (Attack Sustain Release) envelope.

Amplitude envelope release (1ms-16sec) This horizontal slider controls the decay and release portion of (upper) layer's amplitude envelope. This dual function parameter controls both the initial decay phase appearing right after the attack phase, but simultaneously also controls the final decay (often also called release) that appears when a note is released. Setting this parameter low creates very fast, abruptly ending sounds, while using higher values creates lush sounds with long hanging notes.

The Color envelope

The Color envelope is a dedicated AD(Attack and Decay)-envelope that will perform a sweeping of the start and loop-position of the Source waveform, first in a clockwise motion during attack phase, and then in a counterclockwise motion during the decay phase.

Color (0-100%) This parameter control the waveform playback start- and loop-point. This is the offset point from which the playback of the waveform always will begin.

Color movement envelope attack (1ms-16sec) This parameter controls the attack time of the dedicated Color Movement AD (Attack Decay) envelope. The attack time is the timing of the rising portion (attack) of the envelope.

Color movement envelope decay (1ms-16sec) This parameter controls the decay time of the Color Movement envelope. The decay time is the timing of the falling portion (decay) of the envelope.

Color movement envelope amount (0-100%) This parameter controls the amount of color movement inflicted by the dedicated Color Movement envelope. Color movement is the modulation of the start- and playback-position of the Source waveform. The movement inflicted by the Color Movement envelope is added to the offset position set by the Color knob, as well as the modulation added from the Mod Pod (see further description below).

Octave (-1 / 0 / +1) This parameter let the user transpose the Source waveform up or down once octave. Default value is 0 (no transposition).

Pitch (-12 – 0 - +12 semitones) With this parameter, the user can detune the pitch of the Source waveform up or down from the default offset. Maximum values are + and – one octave. Modulate this parameter with the Mod Pod sequencers for creating pseudo-arpeggios (see further description below).

Oneshot (on/off) This parameter turns on the Oneshot playback mode of the Source waveform. In Oneshot mode, sustained loop-functionality is turned off and the waveform is played from the where the current start-position (Color knob + the modulation determines where). The Color-envelope is also disabled in Oneshot mode.

Vibrato (on/off) This parameter turn on the internal vibrato. Vibrato is a low-frequency pitch modulation that emulated the vibrato of a violinist or cellist. When this parameter is engaged, MIDI modulation amount (MIDI controller #1) will control the maximum amount of vibrato (default is 100%).

The Mix Balance functionality

Src Mix This is the mix balance in volume between the upper and lower layer in Parallels. When this knob is turned fully counter-clockwise, only the upper layer will sound. And of course, if this knob is turned fully clockwise, only the lower layer will sound. This described behavior is also reflected in the graphics as the waveform's shown color intensity also will reflect the current state. When a waveform is faded to grey, it is not heard.

The Shaper Section

1. Shaper active on/off
2. Shaper model
3. Hotspot
4. Interactive filter diagram
5. Filter parameters 1-3

Active on/off	This parameter activates and deactivates the Shaper section. When the Shaper is deactivated, its graphics is dimmed indicating that it is no longer heard and the sound from its adjacent Source section is bypassed through to the Effect section.
Type LPG/SVF/RES	This parameter determines which tone shaping model the Shaper section will use. The three types are:
The LPG (Low Pass Gate)	Is a lowpass type of filter inspired by West Coast synthesis, with a variable slew, cutoff and resonance control. While the cutoff and resonance controls works pretty much as on any like other lowpass filter with feedback control, the slew parameter controls the response of modulation applied to cutoff frequency, mimicking the "slop" of a vactrol control.
LPG frequency (20Hz - 20kHz)	This is the cutoff frequency of the LPG. When modulated, the LPG Slew will determine the "sloppiness" its response (see below).
LPG resonance (0 - 100%)	This is the amount of feedback in the LPG Shaper.
LPG slew (0-100%)	This is the amount of "sloppiness" in the cutoff response. It ranges from 20 ms to 2000 ms, minimum to maximum response.
The SVF (State Variable Filter)	Is an OTA-style filter with variable characteristics (lowpass, bandpass, highpas) and separate control over cutoff and the a wild resonance.
SVF frequency (16Hz - 17.6 kHz)	This is the cutoff frequency of the SVF Shaper.
SVF resonance (0 - 100%)	This is the amount of feedback in the SVF Shaper.
SVF type (lowpass - bandpass -highpass)	This parameter set the SVF type used in the Shaper which ranges from lowpass mode with knob at counterclockwise, bandpass with knob at 12 o'clock, to highpass mode at fully clockwise.
The RES (Resonator)	Is a three band variable resonator-bank with control over the center frequency, as well as the tilt and spread of the bands.

RES center freq (159Hz – 2518kHz) This is the center resonator peak cutoff frequency of the RES Shaper mode.

RES tilt (0 – 100%) This parameter set the RES tilt balance used in the Shaper which ranges from high freq boost with knob at counterclockwise to low freq boost at fully clockwise.

RES spread (0 - 100%) This parameter set the spread between the three different bands, from roughly doubling in frequency for each band at fully counterclockwise, to up to eight times the frequency at fully clockwise (x2 - x8).

The Shaper Copy and Swap functionality

The Shaper sections also contains some smartness - you can copy and swap setting between the upper and lower Shaper by clicking on one of the three different "arrow"-buttons (see further description below).

Copy from (arrow on module you want to copy from) Clicking on this arrow on the upper or lower Shaper instantly copies the settings for the Shaper type currently selected, selects the same Shaper type on the other Shaper.

Swap Clicking on this "double arrow" symbol instantly swaps the settings of the upper and lower Shaper. Clicking the same cymbol again swaps back.

Note: No modulation information is copied or swapped with the functions described above. Because of this, settings copied from the upper Shaper to the lower Shaper or vise versa can sound quite different although the same Shaper type and settings are being used. This can of course be cured by manually selecting the same modulation sources, mix and amount.

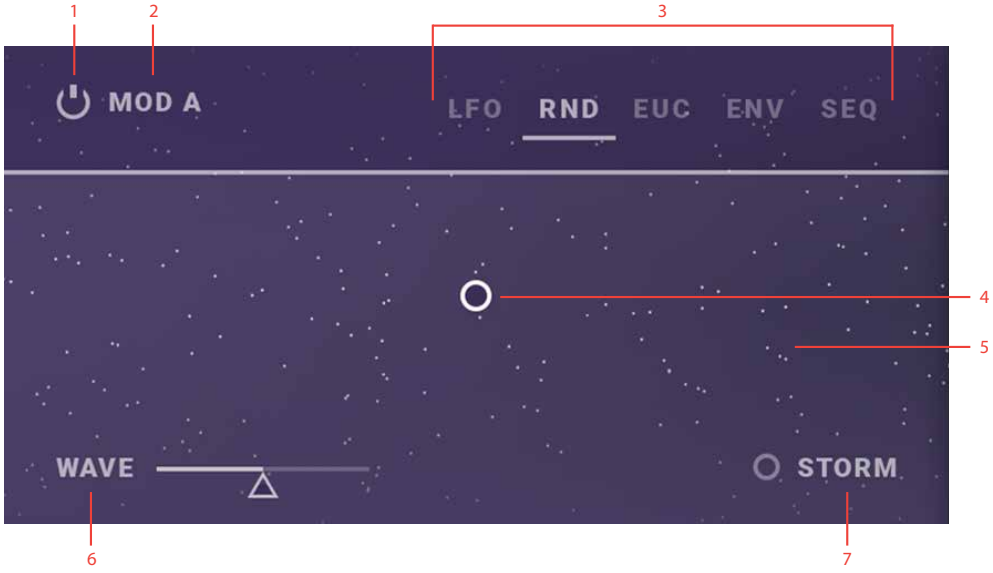
Copy shaper settings to lower.

Swap shaper settings between upper and lower.

Copy shaper settings to upper.

The Mod Pod

The left section of Parallels contains the modulation slots A to D and is called the Modulation Pod or **Mod Pod** for short. Each slot in the Mod Pod contains a configurable modulation source. This means that any of the modulation slots can be assigned to do similar modulation tasks and to be assigned just about anywhere. All slots have their own individual modulation color that is reflected in the assigned modulation.



1. Modulation active on/off
2. Modulation slot A-D
3. Modulation type
4. Hotspot
5. Interactive modulation diagram
6. Modulation parameter 1
7. Modulation parameter 2

Active on/off	This button turns the Modulation Slot on or off. When turned off, modulation is cut from all destinations using it and thus can radically change the sound.
Modulation type (LFO/RND/EUC /ENV/SEQ)	With this type button you can choose which kind of modulation you want the modulation slot to use. Here follows the detailed description of each of these modulation types available in Parallels:

LFO (Low-frequency Oscillator)

An LFO is a low-frequency oscillator - a cyclic rise and fall type of wave oscillation that can be used to modulate your target parameter destination with as slow movement as 0.1Hz (i e 1/10 cycle each second, meaning that cycle time is 10 seconds).

LFO rise (0-100%)	This parameter sets the how much of the LFO cycle that will be the rising portion and thus also the remainder that is to be the fall portion.
LFO shape (0-100%)	This parameter set the shape on the rise and fall period of the LFO curve. It goes seamlessly from inverted exponential (0%) to linear (50%) to exponential at (100%).
LFO DAW speed (4 bars/1 bar/half note/ fourth note/eight note/sixteenth note)	When the LFO sync is turned on, each LFO cycle is quantized into division of a beat ranging from one cycle every 4th bar to one cycle each 16th note.
LFO speed (0.1 Hz – 50Hz)	The LFO speed parameter sets the cycle time of the LFO, from a very slow 0.1Hz to a moderate 50Hz (just above audio-rate).
LFO sync (off/on)	This parameter turn on and off the DAW tempo sync. When turned on, each LFO cycle is quantized into division of a beat (see LFO DAW speed).

RND (Random modulation generator)

The RND is a digital random modulation generator that outputs bursts of random pulses at a regular rate. This means that for each clock-cycle a pulse may of may not appear. The clocked rate in the RND is called "rain" to illustrate the light drops of rain at a low rate, to the heavy rain of a high rate. There's also a low- or highpass filtering of the generated noise as well a internal LFO controlled amplitude-shaping called "wave". The "Storm" option makes the wave go bananas!

RND rain (0-100%) This parameter which is the movement in the Y-axis on RND panel, controls the internal speed of the clocked random source, from the light drops of rain at a low rate (0%), to the heavy rain of a high rate (100%) - white noise. The Y-axis movement can be locked by holding down shift and moving the mouse vertically for only adjustment of the RND rain (clocked speed).

RND noise (lowpass - flat - highpass) This parameter which is the movement in the X-axis on RND panel, controls the filtering of the generated noise, from the heavy lowpass filtered version at 0%, on to flat (no filtering) at 50% and on to heavy highpass filtered at 100%. The X-axis movement can be locked by holding down shift and moving the mouse horizontally for only adjustment of the RND noise (filtered noise).

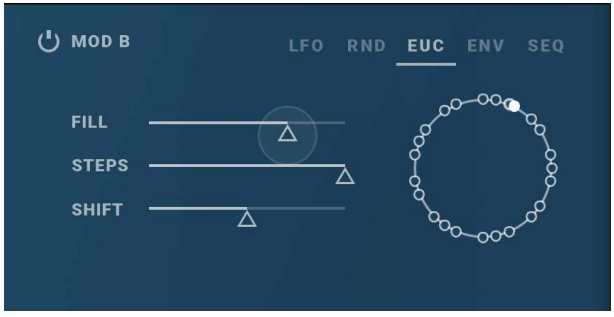
RND wave (0 – 100%) This parameter controls the amount of "breathing" scaling to the modulation amount. This is a cyclic slow amplitude modulation of the filtered random pulse output that can be turned completely off by setting the RND wave to 0%.

RND storm (on/off) This option completely turns off the wave modulation and instead amplitude modulates the RND output with another uncorrelated white noise source.

EUC (Euclidian sequence generator)

The Euclidian sequence generator (EUC) in Parallels can be used to create a number of evenly distributed gates (modulation maximum) over the course of one bar. The number of gates are filled from the first beat clockwise by a percentage and the resulting gates can be shifted around the beat for (nearly) endless array of possibilities. The EUC is always synced to DAW tempo and position, therefore its effect will not be heard when the DAW transport is stopped.

EUC fill (0-100%) This parameter determines how many percentage of the available steps in a bar that will send gates.



For example, in a 16 step sequence a 25% fill will result in 4 distributed gates.

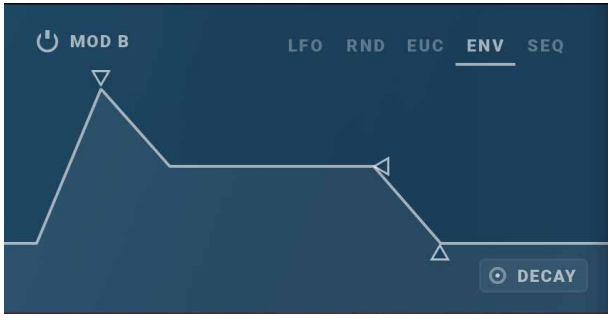
EUC steps (1-32) This parameter determines the number of evenly distributed gates (modulation maximum) over the course of one bar.

EUC shift (0-100%) The EUC shift parameter is used to shift the steps around the perimeter by percentage.

EUC Retrig (off/on) This parameter determine if the EUC cycle will retrigger on next 16th note.

ENV (modulation Envelope generator)

The modulation Envelope generator of Parallels is a ASR or ADSR generator (configured by the ENV Decay on/off button). It can be used to, for example, modulating the filters in the Shaper section, sweeping the waveforms loop start-points and altering the dry/wet level of the effects.



ENV Attack (5ms - 4000ms) This parameter set the attack time of the modulation envelope. It defines the rise time until the envelope has reached its peak. A lower attack value equals a faster rise time.

ENV Sustain (0-100%) The Sustain parameter defines the hold level of the modulation envelope. This is the level that the envelope will rise to from zero through the attack phase that lasts as long as defined by the ENV attack parameter (described above).

ENV Release (50ms - 8000ms) This parameter defines the final fall time of the modulation envelope. It takes from a key is released until the modulation envelope has fallen down to zero again. When the modulation envelope is set to ADSR mode via the ENV Decay option.

ENV Decay (off/on) When this option is turned on, the modulation envelope changes its curve to a ADSR envelope, adding the optional decay stage that will be proportional to, and controlled by the same parameter as the ENV Release (descriped above). In a ADSR envelope, the decay portion describes the fall time from the peak of the envelope, down to its sustain level.

Note: the modulation envelope is the only modulation parameter that is assigned per voice, that is affects each destination polyphonically (true for all destinations except for destinations with the effect section).

SEQ (modulation Sequencer)

The modulation Sequencer in Parallels is designed to provide slewed sequenced modulation or gating information that can be used to modulate different things: pitch, filter cutoff etc. It's speed is a 16th note division set by the DAW sync tempo and its alignment is to transport when DAW is running. The modulation sequencer has two modes: modulation and trig. Where modulation sequencing is stepped level set by each step, the trigged gates of the trig mode is either on or off, and each gate only lasts half of a 16th note (a 32th note). In modulation mode you can easily draw you modulation curves continuously across the 16 steps, and fine-adjustments can be done by pressing and holding CTRL (pc) or CMD (mac) to fine-adjust. This will also prevent more than one step being changed at the same time in the sequencer. Note that the draw functionality can be turned off in the setup menu.

SEQ first step (1-16) This parameter sets the first step of the modulation sequence. If this step occurs after the last step, then the sequence will go backwards. For example, if the first step is 12 and the last step is 8, then the sequence will start at step 12, then play steps 11 through 8 and loop back to step 12 on the next 16th note.

SEQ last step (1-16) This parameter sets the last step if the modulation sequence. If this step is positioned before the first step, the sequence will reverse order as described above.

SEQ slew (0-100%) The slew parameter defines the slew rate of the output modulation from the modulation sequencer. The higher slew value, the slower lag (or glide) is added to the output modulation.

SEQ Trig (off/on) This parameter controls the output mode of the sequencer. Modulation mode (SEQ Trig: off) sequencing is stepped level set by each step, and Trigger mode (SEQ Trig:on) is trigged gates.

SEQ step CV 1 – 16 (0-100%) This is the amount of modulation to be generated by this step.

SEQ step trig 1 – 16 (off/on) In trig-mode, this is the amount (either 0 or 100%) to be generated by the gates of each step. Each gate will last for half a 16th note.

Modulation Routing

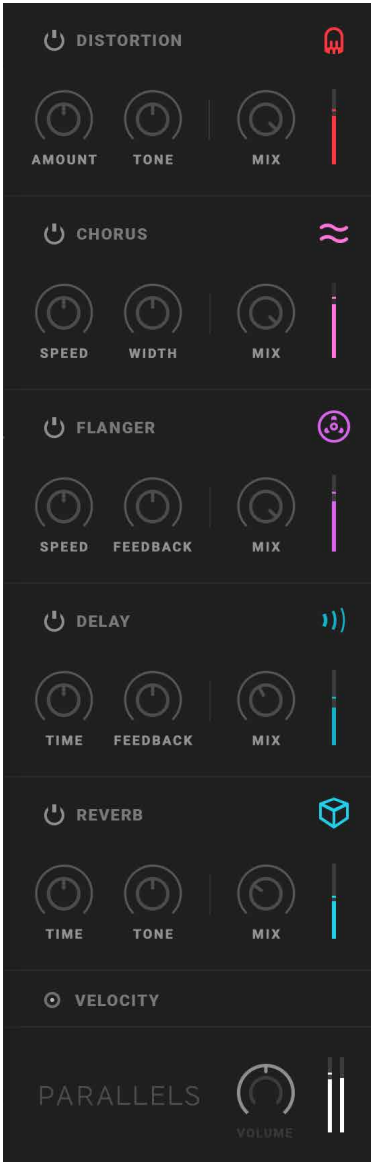
In Parallels, nearly all parameters in Parallels can be modulated from the modulation sources available in the Mod Pod. Click on the outer ring, modulation ring, on a knob to bring up the available modulation sources (A/B/C/D) and the mix balance between them. By clicking on the modulation ring while dragging your mouse/mousepad upwards you will add modulation amount to that destination. The color ring around the destination knob will light up with a color reflecting the modulation amount and mix of the sources chosen to modulate this knob. A 50/50 mix of two sources will result in a white color. There's is also a small indicator on the ring showing the presently applied sum of modulation.

Modulation routed in Parallels is only positive with one exception: Modulation of the to pitch is bipolar, which means that a Source waveform will stay in pitch when modulated with a LFO. This also means that when modulating pitch from the modulation sequencer at full range, nominal pitch will be at 50% on a sequencer step giving the sequencer a modulating pitch range of a total of two octaves.

Note: A few of the parameters cannot be modulated and those are primarily sliders: Color Attack, Color Decay, Color Amount and Octave. Also, oneshot and vibrato on/off options cannot be modulated from the Mod Pod. Change of modulation routings is only reflected in the next triggered note.

The Effect Section

The effects signal flow in Parallels are setup in series, from top to bottom. Each effect has its own dry/wet mix and bypass switch.



Distortion

This effect saturates the signal with a vacuum tube-like flavor.

Distortion Amount (0-100%)	The amount of distortion on the effected signal.
Distortion Tone (0-100%)	Dark or Bright distortion, you decide.
Distortion Mix (0-100%)	This is the dry/wet mix between the dry signal and the effected (distorted) signal. Blend to your preferred ratio.

Chorus

This effect emulates the behavior of a classic stereo-chorus, often found in polyphonic synthesizers of the late 1970s and early 80s.

Chorus Speed (0-100%)	The internal chorus speed.
Chorus Width (0-100%)	The stereo width and amount of the chorus.
Chorus Mix (0-100%)	This is the dry/wet mix between the dry signal and the effected (chorus) signal. Blend to your preferred ratio.

Flanger

A flanger is effect a time-dilation effect where parts of the signal is slightly delayed, feedback and mixed with a constant change of he delayed timed. The results is not so different from a phaser but can still be distinguished as it creates an unlimited series of equally spaced notches and peaks, both harsh and sweet sounding at the same time.

Flanger Speed (0-100%)	This is the flanger internal sweep speed (approximate range 0.5 – 5Hz).
Flanger Feedback (0-100%)	This parameter controls the feedback path of the flanger, but also the stereo width of the sweep; i e the more feedback, the wider flange.
Flanger Mix (0-100%)	This is the dry/wet mix between the dry signal and the effected (flanged) signal. Blend to your preferred ratio.

Delay

This delay is an echo effect quite similar to the Filter echo of Softube Heartbeat. It features both a set delay-speed on the first half on the speed knob. But also a variable delay-speed set as division of a beat synced to your DAW, on the second half on the same knob.

Time (1-900ms, 1/16, 1/32, 1/16, 1/16+, 1/8T-, 1/8T, 1/16D, 1/8, 1/8+, 1/4T-, 1/4T, 1/8D, 1/4, 1/2T, 1/4D, 1/2)	This parameter determines the delay time of the echo; i e how long time passes between each delay "hit". On the left half of the Time knob, the range is from 1 to 1000 milliseconds. On the right side, the delay time can be set in divisions of the DAW tempo, ranging from 1/64th to 1/2 of a beat. The latter is useful for setting the delay to act in time with your song.
Feedback (0-100%)	Is set the amount of feedback, how many delay repeats there will be. It ranges from one repetition to roughly 10 repetitions at full feedback.
Delay Mix (0-100%)	This is the dry/wet mix between the dry signal and the effected (delayed) signal. Blend to your preferred ratio.

Reverb

The Reverb in Parallels is Softube custom reverb algorithm tailored to suit the needs of a synth sound. It creates everything from a short slap-back room to a vast synthesized textured space. Tweak it according to your needs.

Time (0-100%)	This parameter changes many things at once about the reverb. Low values makes the pre-delay more apparent, with less feedback and shorter times in the feedback loop. 20-40% creates a small space and with values higher than 50% a very large space is created by the reverb.
Tone (0-100%)	This parameter sets the filtering of the feedback networks in the reverb. While 50% is a flat filter topology, lower values below 50% will dampen higher frequencies in the reverb (dark reverb). Values above 50% will dampen lower frequencies in the reverb (brighter reverb).
Reverb Mix (0-100%)	This is the dry/wet mix between the dry signal and the effected (reverb) signal. Blend to your preferred ratio.

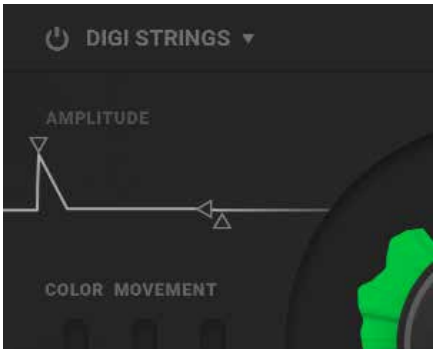
Setup

By clicking on the Setup window tab in the lower right corner of Parallelss widow, a screen of global options will be displayed. Many of the options requires shutting down and opening Parallels again (good to know). The setup options are as follows:

Warn when deleting presets from preset collection	Turns off or on the dialogue appearing when deleting user presets in the preset collection. Factory presets cannot be deleted, although they can be filtered out (not shown).
Warn when overwriting presets in preset collection	This option turns off or on the dialogue appearing when overwriting user presets in the preset collection.
Use Open GL graphics	Turns off or on Open GL graphics acceleration. This is an option that affects all Softube Plugins and if selected, a DAW restart is required before change of this option is active.
Color Blind Mode	This option enables or disables colors specially adapted for the colorblind. Close and open the Parallels GUI for the change to have effect.

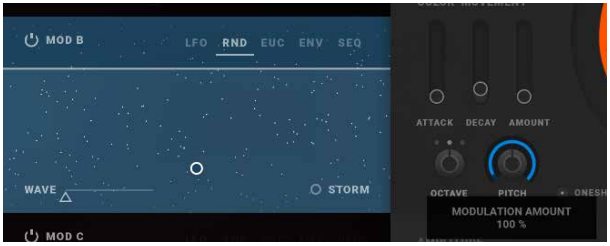
Show Colors in Menu	This option enables or disables colors in the wave menu reflecting the colors of the waveforms already before they are selected. Close and open GUI again for the change to have effect.
Always use smaller GUI	This option forces Parallels's rescaler to always open the plugin in a smaller window. Close and open the Parallels GUI for the change to have effect.
Show tooltips	Turns on or off the tooltips (screen text overlay for parameters). Close and open the Parallels GUI for the change to have effect.
Show value display	Turns on or off the value display for edited parameters in the lower left corner. Close and open the Parallels GUI for the change to have effect.

Tips and tricks



Use the Amp Envelope to change the dynamic character of any sound. Pictured above is the difference on the Amp Envelope between a plucked string sounds and slow, bowed stringed sound.

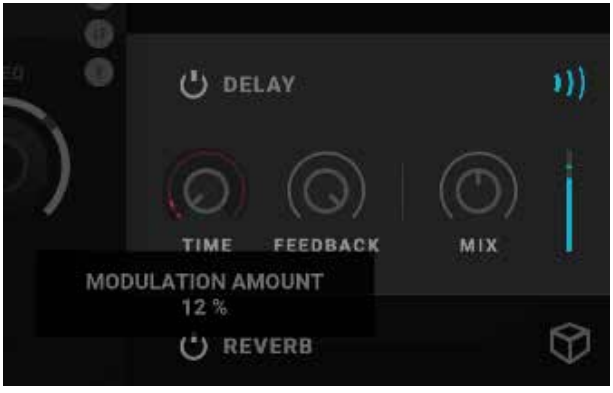
Most of the different Source waveforms have change of harmonic content when sweeping the Color knob clockwise. Make use of these harmonic changes either by using the dedicated Color Movement Envelope or by assigning a modulation slot to create a harmonic change while playing. And of course you can also do both at once. Pictured above is how a LFO in Mod Pod C is assigned to sweep the lowpass filter in the "Ana Saw LP" Source Waveform.



Use the RND to create random octave jumps with the RND Wave parameter set at 0. Use RND modulation on full 100% amount to make the RND pulses to create moderatly slow random octave jump.

You can modulate the internal LFO speed of the flanger with a modulation LFO by assigning a Mod Pod containing the LFO to the LFO speed knob of the Flanger (as pictured above).

Total polyphony is shared between upper and lower layer. Double your polyphony by turning off one layer if you do not need it.



Create your own unusual flange effect by assigning a modulation LFO to speed knob of the Delay effect (as pictured).

Create a tremolo by using a modulation LFO to modulate mix with one Source turned off.

Another way of creating a tremolo effect is by setting a late start in the Source Wave and assigning a modulation LFO to modulate playback.

Credits

Johan Antoni – Initial concept, multi sample sound-recording and library, testing and presets. **Kristofer Ulfves** – Project lead, sound design, presets, user manual. **Björn Rödseth** – Tech lead, programming. **Kim Larsson** – DSP modeling, programming. **Erik Sight** – programming. **Patrik Holmström** - programming. **Jacopo Lovatello** – programming. **Filip Thunström** – programming, preset conversion. **Tord Jansson** – programming and GUI engine customization. **Oscar Öberg** – Mentoring, programming. **Arvid Rosén** – Mentoring, programming. **Manuel Colomb** – GUI design. **Joe Lawton** – Testing, presets. **Henrik Johansson** – Testing, presets. **Fredrik Mjelle** – Testing, presets. **Christoffer Berg** – Testing, presets. **Maxus Widarsson** – Deep testing, Qualification. **Johan Bremin** – Deep testing, Qualification. **Sven Bornemark** – Testing. **Fanny Hökars** – User Manual layout.

Appendix

Waveforms list

Cathegory	Index no	Name	Description clockwise harmonic change
Digital Synth	001	Digi Guitar	Synth-guitar sound
	002	Digi Noisy Seagulls	a slowly swelling synthetic sea-gull sound
	003	Digi Strings	A somewhat brassy synth-string sound
	004	Digi Meta 1	Digital wave that propagates from a soft to a harsh sound
	005	Digi Meta 2	Digital wave that evolves from a soft pad into a harsh piercing sound.
	006	Digi Meta 3	Digital chunky chord sound that evolves into a huge space.
	007	Digi Meta 4	Digital guitar into added harmonic content.
	008	Digi Meta 5	Digital plucked string sound extended into a held tone warped into distortion.
	009	Digi Meta 6	Digital percussive looped sound into warped distortion
	010	Digi Meta 7	Digital guitar into warped distortion.
	011	Digi Superwave	Static sawtooth into a detuned swarm of supersaw.
	012	Digi Table 1	Swept digital wavetable with different harmonic content.
	013	Digi Table 2	Another swept digital wavetable with different harmonic content.
	014	Digi Table 3	Swept digital wavetable from even to odd harmonics.
	015	Digi Table 4	Digital additive synthesizer style wavetable.
	016	Digi Table 5	Swept formant wavetable.
	017	Digi BP Amped	Digital distorted triangle wave through bandpass.
	018	Digi LP Decimated	Digital distorted lowpassed triangle wave.
	019	Digi LP Dist Overload	Another digital distorted lowpassed triangle wave.
	020	Digi Tri Meta 1	Digital choir into distortion.
	021	Digi Tri Meta 2	Digital female choir into distortion.
	022	Digi Tri Meta 3	Digital voice into distortion.
	023	Digi Wave	Digital wave with filter sweep.
Analog Synth	024	Ana Hoover	Classic hoover sound from the analogue classic.
	025	Ana Sweep	Filter LP sweep sound from an analogue poly classic.
	026	Ana Saw BP	Sawtooth bandpass-sweep from a vintage analogue synth.
	027	Ana Saw HP	Sawtooth highpass-sweep from a vintage analogue synth.
	028	Ana Saw LP	Sawtooth lowpass-sweep from a vintage analogue synth.
	029	Ana Squ BP	Squarewave bandpass-sweep from a vintage analogue synth.
	030	Ana Squ HP	Squarewave highpass-sweep from a vintage analogue synth.
	031	Ana Squ LP	Squarewave lowpass-sweep from a vintage analogue synth.
	032	Ana Tri BP	Trianglewave basspass-sweep from a vintage analogue synth.
	033	Ana Tri Hp	Trianglewave highpass-sweep from a vintage analogue synth.

	034	Ana Tri LP	Trianglewave lowpass-sweep from a vintage analogue synth.
	035	Vintage Saw	Classic vintage sawtooth sound through lowpass sweep.
	036	Vintage Saw Feedback	Saturated classic vintage sawtooth sound through lowpass sweep.
	037	Vintage Saw Res	Classic vintage sawtooth sound through resonant lowpass sweep.
	038	Vintage Squ	Classic vintage squarewave sound through lowpass sweep.
	039	Vintage Squ Res	Classic vintage squarewave sound through resonant lowpass sweep.
	040	Vintage Tri	Classic vintage trianglewave sound through lowpass sweep.
	041	Vintage Tri Feedback	Saturated vintage trianglewave sound through lowpass sweep.
	042	Vintage Saw LP Amped	Amped vintage sawtooth sound through resonant lowpass sweep.
Combo	043	Combo 1	Combination of waveforms with an increased harmonic complexity.
	044	Combo 2	Combination of waveforms with an increased harmonic complexity.
	045	Combo 3	Combination of waveforms with various pulse wave content.
	046	Combo 4	Combination of squarewaves with different tuning.
	047	Combo 5	Combination of waveforms morphing into each other.
Chords	048	Chords 1	A minor 3rd interal tuned up to a 5th
	049	Chords 2	Unison tuned up to a minor 3rd
	050	Chords 3	Minor 3rd with increased FM modulation.
	051	Chords 4	Minor 3rd stab with increased assymetric triggering.
	052	Chords 5	Minor 3rd stab with increased assymetric triggering.
Drones	053	Drone Combo 1	Atonal drone with increased FM amount.
	054	Drone Combo 2	Atonal feedback sound.
	055	Drone Combo 3	Atonal cyclic silent drone fade up.
FM	056	FM Mallet	Classic icy FM Mallet sound.
	057	FM 1	4 operator FM sweep with feedback.
	058	FM 2	Even broader 4 operator FM sweep with feedback.
	059	FM 3	4 operator FM sweep, different pitch relations
	060	FM 4	Dubstep style FM sweep
	061	FM 5	Harsh FM sweep
	062	FM 6	Broader and harsher FM sweep
Physical Mod	063	Phys Mod Hybrid	Woodwind pulse, inverse amplitude sweep
	064	Phys Mod 1	Hollow body wave
	065	Phys Mod 2	Bowed string wave
	066	Phys Mod 3	Modal string, inverse amplitude sweep
	067	Phys Mod 4	Rubber into odd harmonics sweep

	068	Phys Mod 5	Rubber into FM sweep
Stacked	069	Stack 1	Oscillator stack with pitch variations in the beginning and end
	070	Stack 2	Sine stack through waveshaper
	071	Stack 3	Bounce stack with increased rate
	072	Stack 4	Pulse stack
	073	Stack 5	Square to Triangle shape stack
	074	Stack 6	Triangle to pulse shape stack
	075	Stack 7	Sinusoid sub stack to FM transformation
	076	Unison 1	Unison squares with added pitch modulation
	077	Unison 2	Unison tilt modulated saw/triangle
	078	Unison 3	Unison pulse with added pitch modulation
	079	Unison 4	Unison square-sine with added pitch modulation
Distorted	080	Thrashy 1	8-bit style pulse trash wave
	081	Thrashy 2	8-bit style pulse trash wave
	082	Thrashy 3	Lo-fii FM with sync portion
Environment	083	Ebisu 1	Enviromental sounds from Ebisu area blended with sines
	084	Ebisu 2	Enviromental sounds from Ebisu area
	085	Ebisu 3	Gong from Ebisu with freezed looping
	086	Ebisu 4	Power station in Ebisu
Chaos	087	Chaos 1	Chaotic analogue synth sweep
	088	Chaos 2	Chaotic analogue synth sweep
	089	Chaos 3	Particle synth sweep
	090	Chaos 4	Sine noise decay
	091	Chaos Phys Mod 1	Chaotic physical modelling synth sweep
	092	Chaos Phys Mod 2	Particle string into square
	093	Chaos Phys Mod 3	Irregular string model
	094	Chaos Phys Mod 4	Particle string into pad
	095	Chaos Phys Mod 5	Atonal string, inverse amplitude sweep
	096	Vintage Chaos 1	Westcoast Chaos
	097	Vintage Chaos 2	Westcoast Chaos

41 Passive Equalizer

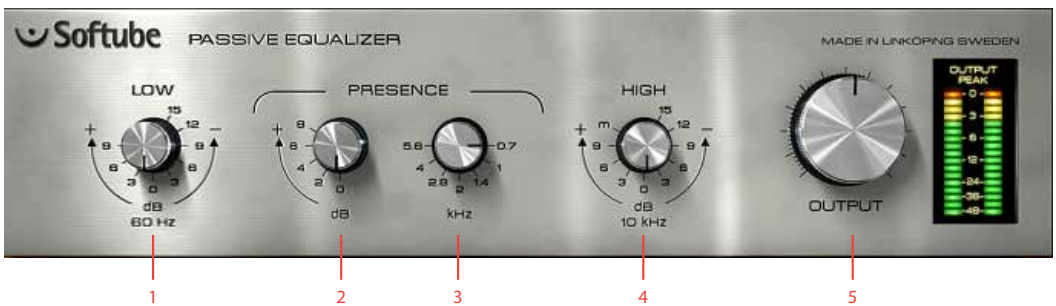
For more info about this product, please see the chapter about the Focusing Equalizer.

The Passive Equalizer, modeled from a Neumann PEV 930 console equalizer, has a very wide sound with a warm low end and a sparkling top. Excellent for adding that extra sparkle a track or a mix might need. The Mid set at 700Hz will give you a fat 70's sounding snare drum. Technically speaking, the High and Low filters are more of a shelving type than a bell filter, and the Mid filter is widest at low frequency settings, while getting more sharp for higher frequencies. Since this is a passive circuit, all knobs will interfere with each other and changing something in the treble might do something in the bass or mid as well.

The smooth and sparkling high boost filter is great for making up for a dull and lifeless track.

Knobs

Low Cut/Boost This knob will either cut or boost the low frequencies. The original unit is labeled as if it boosts or cuts with 3dB steps at 60 Hz, but in reality it changes the frequency of a shelving filter so that the attenuation/gain at 60 Hz will change. It would be more correct to label it with a fixed gain and a variable frequency, but we decided to leave it as it was on the original unit. And due to the non-ideal components in the unit, it does more things than that but that's too technical to put in a user's guide...



1. Low gain
2. Mid boost
3. Mid frequency
4. High gain
5. Output volume

Presence (Gain Control)	Use this to set the gain of the presence circuit in 2 dB steps (from 0 to +8 dB)
Presence (Frequency Selector)	Set the center frequency of the presence circuit (0.7, 1, 1.4, 2, 2.8, 4 and 5.6 kHz). The center frequency and gain have a tendency to move if you increase the **High** knob...
High Cut/Boost	Boost or cut the high frequencies. Just like the **Low** knob, this is more of a shelving type filter than a bell type. Boosting the **High** will change the character of the **Presence**.
Output	Output volume. Goes from nothing to +12 dB. At the 12 o'clock position, it's set at 0 dB, and going clock wise will increase the output volume by +2 dB for each mark.

The Presence filter is widest at low frequency settings (0.7 kHz) while getting sharper at higher settings (5.6 kHz). The 0.7 kHz setting is excellent for getting a fat snare drum sound.

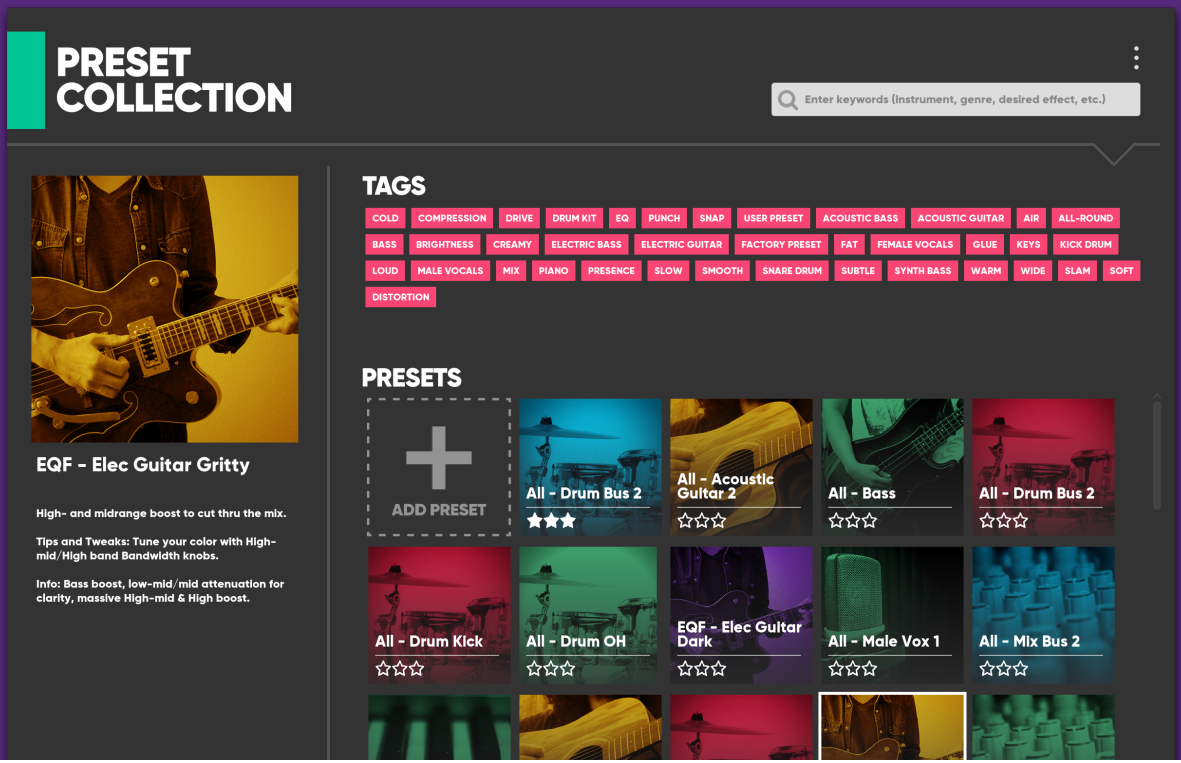
Buying Recommendations

If you like the sound of the Passive Equalizer and would like to get your hands on the real stuff (which is an excellently built and beautifully crafted unit), you should look for a Neumann PEV 930-00 console equalizer. It's a fantastic sounding unit.

ALL PRODUCT NAMES USED ARE TRADEMARKS OF THEIR RESPECTIVE OWNERS, AND IN NO WAY CONSTITUTES AN ASSOCIATION OR AFFILIATION WITH SOFTUBE. THE NEUMANN AND PEV TRADEMARKS ARE SOLELY USED TO IDENTIFY THE PRODUCTS WHOSE SOUND WAS STUDIED DURING THE SOUND MODELING DEVELOPMENT OF THE PASSIVE-ACTIVE PACK. NEUMANN® IS A REGISTERED TRADEMARK OF GEORG NEUMANN GMBH.

Credits

Niklas Odelholm – modeling and graphics design. **Torsten Gatu** – concept. **Oscar Öberg** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – graphics rendering. Thanks to **Stefan Fandén** and the crew at Deluxe Music for letting us borrow the gear!

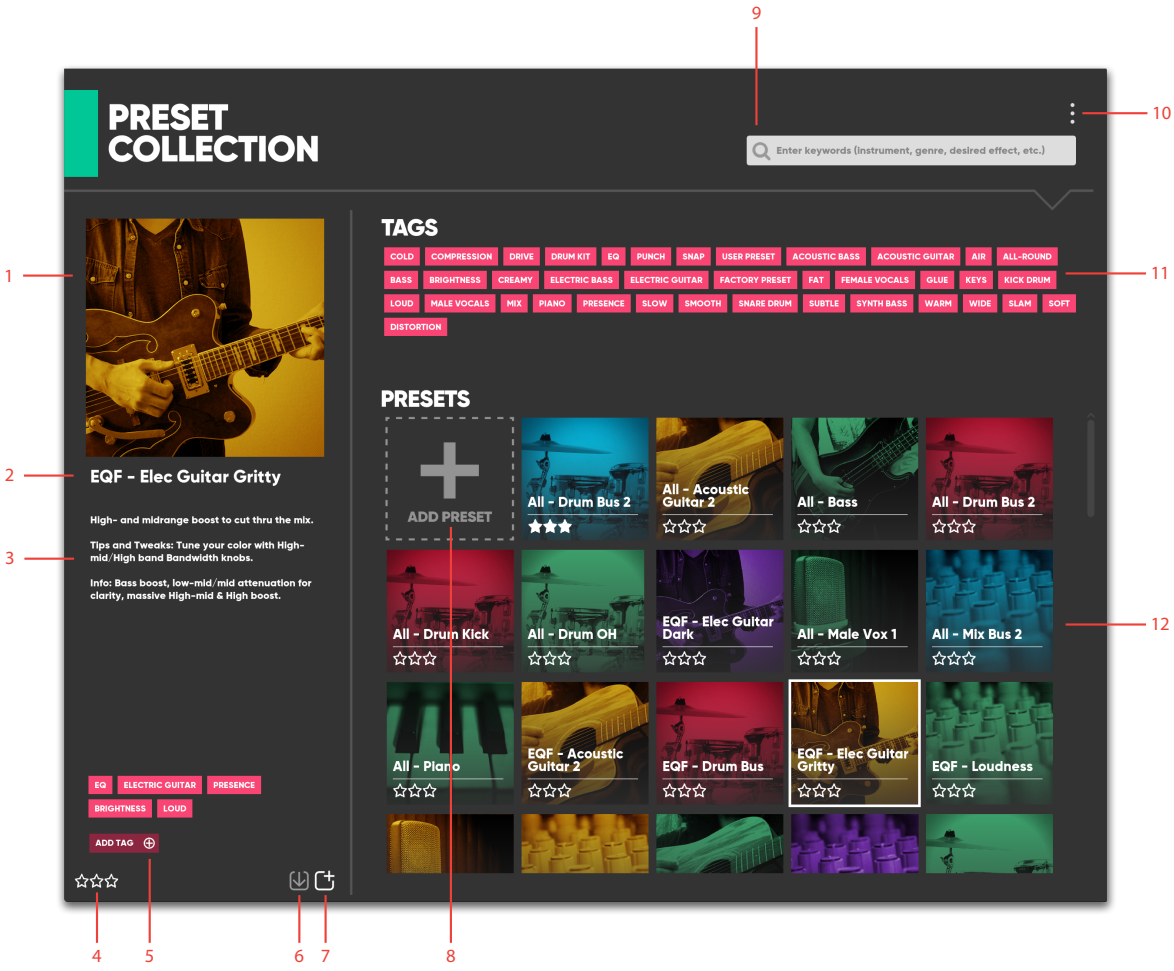


42 Preset Collection

What is it for?

In Softube plug-ins, presets are organized in the Preset Collection. Each plug-in has its own collection of presets and you can reach it by clicking the **PRESETS** button in the plug-in UI.

Overview


PRESET COLLECTION
Enter keywords (instrument, genre, desired effect, etc.)
TAGS
COLD COMPRESSION DRIVE DRUM KIT EQ PUNCH SNAP USER PRESET ACOUSTIC BASS ACOUSTIC GUITAR AIR ALL-ROUND
BASS BRIGHTNESS CREAMY ELECTRIC BASS ELECTRIC GUITAR FACTORY PRESET FAT FEMALE VOCALS GLUE KEYS KICK DRUM
LOUD MALE VOCALS MIX PIANO PRESENCE SLOW SMOOTH SNARE DRUM SUBTLE SYNTH BASS WARM WIDE SLAM SOFT
DISTORTION
PRESETS
ADD PRESET
All - Drum Bus 2
All - Acoustic Guitar 2
All - Bass
All - Drum Bus 2
All - Drum Kick
All - Drum OH
EQF - Elec Guitar Dark
All - Male Vox 1
All - Mix Bus 2
All - Piano
EQF - Acoustic Guitar 2
EQF - Drum Bus
EQF - Elec Guitar Gritty
EQF - Loudness
EQF - Elec Guitar Gritty
High- and midrange boost to cut thru the mix.
Tips and Tweaks: Tune your color with High-mid/High band Bandwidth knobs.
Info: Bass boost, low-mid/mid attenuation for clarity, massive High-mid & High boost.
EQ ELECTRIC GUITAR PRESENCE
BRIGHTNESS LOUD
ADD TAG
1
2
3
4
5
6
7
8
9
10
11
12

1. Image
2. Preset name
3. Description
4. Stars
5. Add Tag button
6. Update button
7. Create a new preset button
8. Add preset button
9. Search bar
10. Three dot menu
11. Tags
12. Presets section

How do I use it?

To load a preset from your collection, select it in the **Presets** section. This will change the plug-in settings, so be careful about overwriting settings that you may want to keep.

If you are looking for something specific, choose some words in the **Tags** section first, to narrow down the list of presets.

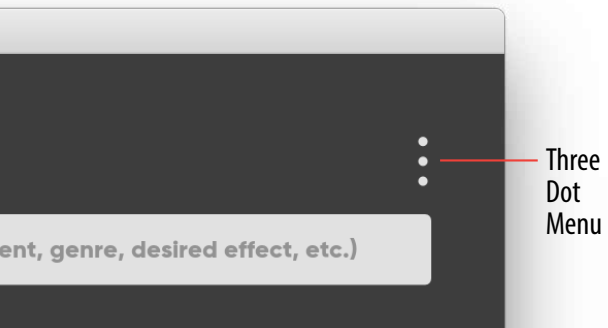
Mark presets that you like and think you want to use again with **stars** in the bottom left corner and they will show up first in the Presets list.

If you want to save your own preset in your collection, press the **Add Preset (+)** button in the **Presets** section.

Choose a name and maybe a photo (recommended size 548x548), description, and some tags. Then click the **Create a new preset button**.

Tip: Tag the presets you create with your name to be able to find them easily later.

Import a preset or a bundle of presets to your collection by using the **three dot menu** in the upper right corner. You can only import presets for the plug-in you're currently running.

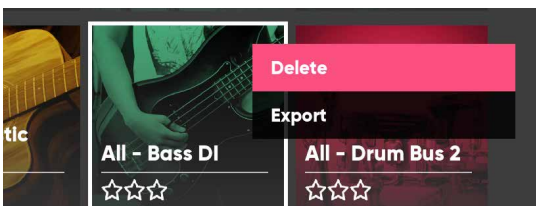


Send presets to friends, colleagues, or a second computer by using one of the **Export** functions in the three dot menu. You can export single presets (in .softubepatch format) or all of the presets in the current search result as a preset bundle (in .softubebundle format).

Tip: Add the same tag to all presets that will go into your preset pack, select it in the Tags section, and click Export Selected Presets in Search in the three dot menu.

Update a preset by loading it, tweaking some plug-in settings, tags, etc. and pressing the **Update** button. For factory presets only the number of stars can be updated, but you can create a user preset based on a factory preset by clicking the **Create a new preset button**.

Delete a preset by right-clicking on it and clicking **Delete** in the popup menu that appears. Factory presets can't be deleted.



How were the factory presets designed?

All of the Softube plug-ins come with factory presets and they were created with common philosophy. There's no need for your own presets to use that same philosophy and we would rather encourage that users experiment with the pretty open-ended system we've created. We did think it would be nice to make the experience of using factory presets consistent though.

Description

These always describe the purpose of the preset ("What is it for?"), often have notes on usage ("How do I use it?"), sometimes tips and tweaks ("How can I take it further?"), and sometimes additional info ("How was it made?").

Tags

The overall tag structure in Preset Collection is flat and anything can be a tag, but for the factory presets we stuck to four tag types:

1. What type of processing is it doing? Compression, Reverb, Synthesizer, ...

2. What's the typical use? Male Vocals, Lead Synth, …

3. How does it sound? Dark, Soft, Heavy Dist, …

4. Who made it? We are proud of the artists we've worked with for some of our presets and believe in credit where credit is due.

Image

To make it easy to find the right preset, the factory preset images are always a combination of two things. The first is a photo, which describes the typical use (microphone for vocals, etc) or the type of processing. The second is a color describing the sound character:

Red: Hard, aggressive. An over the top sound.

Green: Nice, soft, warm. A normal sound.

Yellow: Bright, open, light. A normal sound.

Purple: Deep, dark, ominous. Not so normal.

Blue: Metallic, high pitched, uneven harmonies. Not aggressive though, which would be red.

MODULAR READY

43 Saturation Knob For Modular

Saturation Knob is a modeled output distortion that can be used anywhere you need some grit. Use it to fatten up bass lines, add some harmonics and shimmer to vocals, or simply destroy your drum loop.

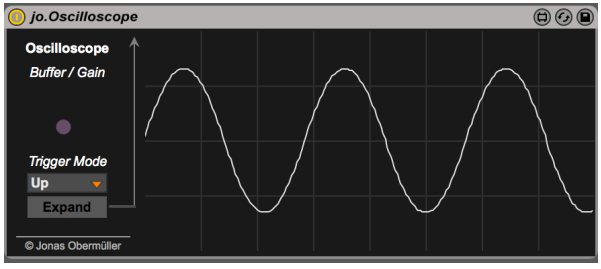
The three modes, Keep High, Neutral and Keep Low, give you three kinds of distortion characters. If you want to squash a drum loop, but at the same time keep the bass drum fairly intact, you can for example use the Keep Low mode. Vice versa you can use the Keep High mode to keep the hi-hats intact but distort the lower end of the bass-drum and snare.

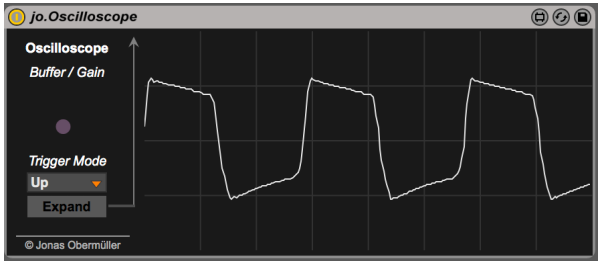
Overview

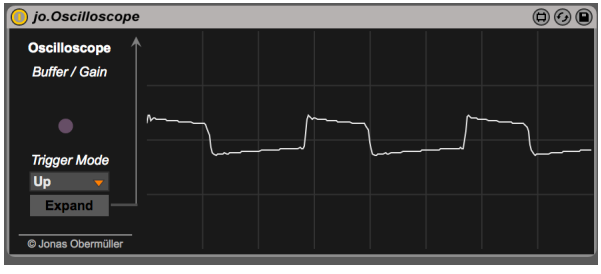
1. Saturation knob
2. Saturation type
3. Saturation CV input
4. Left and right inputs
5. Left and right outputs

Parameters

Saturation knob
This knob is THE actual saturation knob from which the plugin got its name. It controls the saturation amount, clipping the waveform and ear pleasingly squishing as saturation is increased. When set fully anti-clockwise no saturation takes place and turning the knob obviously increase saturation.







Saturation Type
This switch decides the balance of the saturation:

Keep Low – Saturation is applied on the high end of the signal fed through.

Neutral – All frequencies are equally distorted.

Keep High – Saturation is applied on the low end of the signal fed through.

Inputs

Saturation CV input
This input is external CV control of the Saturation amount. CV input here will be added to the amount set by the saturation knob.

Left input
Left of the stereo inputs, insert your signal to distort here. This input is normalized to both stereo outputs if no signal is inserted into the right input

Right input
Right of the stereo inputs, insert your signal to distort here.

Outputs

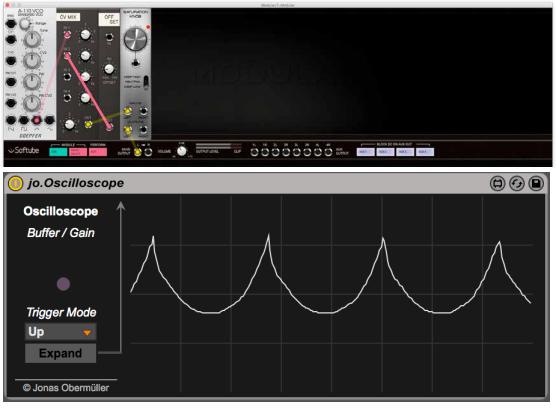
Left input
Left of stereo outputs, this output is a mono mix of both stereo inputs distorted through the Saturation Knob module if no other signal is inserted into the right output.

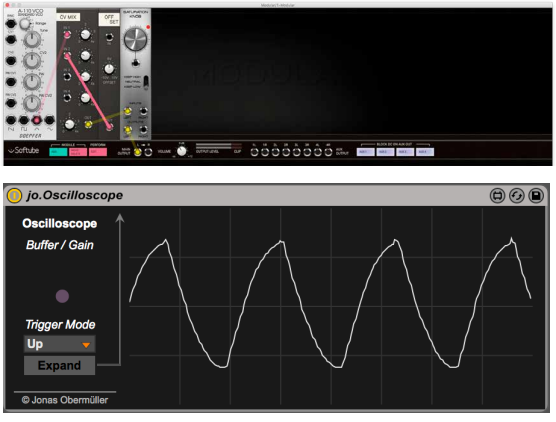
Right input
Right of the stereo outputs.

The Saturation Knob module in use

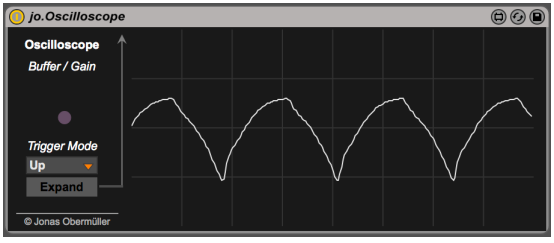
Using the Saturation Knob module in the feedback line of the A-108 VCF or in a delayed feedback line to create gritty cool echoes is highly recommended.

The Saturation knob mode can also be used as a crude wave-shaper for waves that lack a rich harmonic register such as sine and triangle waveforms. Use offsets and/or CV-mixers to control the asymmetry of the clipping.





Use case 2 (continued).



The saturation becomes more pronounced if the input signal contains a lot of uneven harmonics. Try for instance to patch two sine-oscillators through an audio mixer and then into the Saturation Knob module, and then set their relative pitches to an uneven frequency relationship (for example 65.4hz and 177.6hz, a ratio of 2.71). Now, when the saturation is increased, inharmonic sidebands appear as a very harsh pronounced distortion in the audible signal. When changing the second oscillator frequency so that the relative pitch relationship is even (for example 65.4hz and 98,1hz, a ratio of 1.5) the sidebands created by the distortion have a harmonic relationship and sound much more pleasing. This is the same principle on which "power chord" theory with electric guitars is built.

44 Spring Reverb

Introduction

THE SPRING REVERB WAS AN essential part of almost all albums produced before the age of digital processing. With a sound that doesn't sound at all like a reallife reverberation, it has made its way into the music production halloffame just because of its peculiar character. There is really no reverb that sounds like a spring reverb, and if you want to get a vintage vibe on your production, that's about the only way to go.

The Spring Reverb plugin can for example be used as an insert effect together with the Brown amp from Vintage Amp Room in order to get a really bluesy sound, or as a send effect on vocal tracks to get that vintage vibe. The Springs and Tension parameters let you morph between a typical guitar amp reverb and a smoother studio reverb, thus making the Spring Reverb plugin a very versatile tool in your studio rack.

As with all our products, Spring Reverb is the result of exhaustive research. It was developed after nearly a year's analysis of the mathematics and mechanics of spring reverbs. New simulation techniques had to be developed in order to simulate the springy characteristics of the reverb, and we're proud to say that we found what makes a spring sound springy. The user interface was designed to be as simple as possible, with a small set of parameters that make a difference. The Tension and Springs parameters are additions that we didn't find on any real reverb, but that we just couldn't live without.

1. Mix (Dry/Wet)
2. Bass
3. Treble
4. Springs
5. Tension
6. Shake!

User Interface

The Spring Reverb has the following controls: **Mix**, **Springs**, **Tension**, **Shake**, **Bass**, and **Treble**. The following pages will give you a brief description of all the controls.

Mix Controls the balance between the reverb and the direct signal. Outputs only the original signal when set to DRY, and only the reverb effect when set to WET.

Set this control to blend the wanted amount of reverb with the dry signal. Make sure that you set **Mix** fully clockwise on WET every time you use Spring Reverb as a send effect.

Springs Controls the number of springs in use. There are three sets of springs in the reverb unit and you can choose how many of the sets you would like to use.

Set the control to ONE for a more pronounced springy sound and to THREE for a smoother sound.

Together with the **Tension** slider, this control is an excellent tool to change the overall character of the reverb.

With **Springs** set to TWO and **Tension** in the left most position, you will get a typical Accutronics® Type 4 sound, as used in many American guitar amps.

Shake BOOOM! Since we don't want you to smack your computer to get that thunderous springs-being- shaken sound, Spring Reverb features a slider that can be used to accomplish the same thing safely. Fully automatable of course.

Be aware that when the slider is moved to a position other than halfway between far left and right, the reverb will go "spoing", so storing presets with this knob set to something other than in the middle, it will produce a (perhaps unwanted) noise when the preset is loaded. For this and other reasons, when the slider is adjusted in the GUI it will spring (ha!) back to the middle when the mouse button is released. This does not work when the parameter is being controlled by a MIDI surface or other controller.

The **Bass** and **Treble** controls do not affect the sound of the shake effect, since they are applied before the signal goes into the reverb tank.

For the most violent shake effect: set **Tension** to the left and **Springs** on three before you use the **Shake** parameter.

Tension Tunes up or down the tension off all three springs. Adjusting the **Tension** parameter will change the sound of the strings from slow and smooth (left) to quick and harsh (right). The idea behind this parameter is that you are able to get the sounds from many different reverb tanks by just changing the tension and the number of springs. A cheap guitar reverb tank will usually have a high tension and two springs, while a more expensive reverb tank might have three springs and a lower tension.

Bass and Treble Controls equalization of the wet signal. In general you would need to lower the **Treble** if you use the reverb on percussive sounds, while too much **Bass** will make the reverb sound very dark and take up a lot of energy in the mix.

Tweaking Tips

There are a couple rules of thumb to achieving the sound you desire with Spring Reverb.

Mix

Mix is without doubt the setting that will have the biggest impact on the overall sound you achieve and probably the one that is hardest to master.

First of all, when used as an insert effect, the **Mix** knob will typically be set between 0 and 20%. It is easy to drench your recording with a reverb that is as characteristic as Spring Reverb.

When used as a send effect, the **Mix** knob should be set to 100%, but the send/return volumes of your DAW should be kept low. Easy does it!

Controlling the Character

In terms of character, Spring Reverb goes from vintage grit, full of character, to semi smooth, still with character but toned down.

The grittier sounds are for example great with electric guitars, electric pianos, and vocals. Setting the **Springs** knob low and **Treble** high will bring out the spring characteristic, producing more pronounced echoes. Then **Bass** can be used to shape the overall frequency characteristic and **Tension** to control the length of time it takes for the reverb to stop ringing.

For more percussive sounds, such as drums or a full mix, the character will probably need a bit of toning down. Bring the **Springs** knob all the way up to let the echoes of the three springs intermingle and create a more solid reverb tail. Then reduce **Treble** to decrease the metallic character and finally do final tweaks with **Tension** and **Bass**.

Using Spring Reverb with Vintage Amp Room

Spring Reverb is an excellent companion to the Brown amp in Vintage Amp Room. In the real amplifier, the spring reverb is placed in between the preamp and power amp. With the Spring Reverb plugin, you can use it both as an insert *before* Vintage Amp Room and as an insert *after* Amp Room and get very different results.

Placing Spring Reverb before Vintage Amp Room

If you place Spring Reverb before Vintage Amp Room, the reverb effect will be distorted in the same way as the guitar signal, and you will usually need to lower the **Mix** knob and use a bit less **Treble** than if you place it after the plugin. This placement will often make the reverb sound more authentic, although it is a bit harder to dial in.

Placing Spring Reverb after the Vintage Amp Room

This is the "normal" way to use Spring Reverb. All presets have been tailored to suit this use. Just place Spring Reverb (preset: "GUITAR DIRTY") after Vintage Amp Room (preset: "BROWN BLUES") and you'll get the most authentic blues sound available from a plug-in. Don't forget to use a beaten up ol' guitar with single coil mics.

Chorus-y Effect

It is very easy to get a very special chorus effect if you automate the **Tension** knob. Since many hosts (such as Cubase and Pro Tools) have the ability to write in automation curves that are sinusoidal or triangle shaped, it is very easy to get an unique sounding chorus effect from Spring Reverb.

Credits

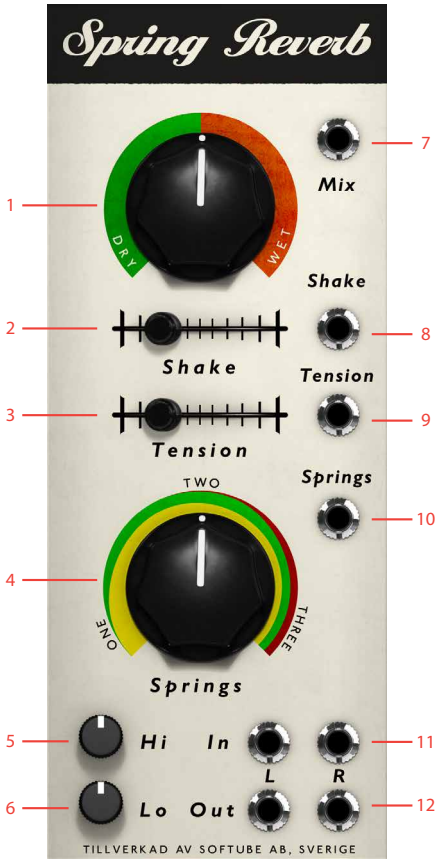
Peter Möller – research and mathematical modeling. **Torkel Svensson** – research and mechanical analysis. **Oscar Öberg** – modeling and implementation. **Torsten Gatu** – framework programming. **Niklas Odelholm** – framework programming and graphic design. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering and graphics.

MODULAR READY

45 Spring Reverb For Modular

Back in the 1930s, Laurens Hammond had started developing and selling his now famous Hammond electro-mechanical Organ. While he had bought a small church organ, having it installed in his living room, he soon found out that it didn't sound as impressive without the reverberation of the church hall. To cure this, he borrowed technology from Bell Labs who had developed a device to emulate the artificial delay in long-distance calls using wire and springs. Hammonds new tweaked version of Bell Labs device was the first real artificial reverberation unit and it was huge. Eventually, in the 1960s, the spring reverb units had shrunk enough to fit into guitar amplifiers and so they also were incorporated in the early synthesizers such as the EMS VCS3 (1969), ARP 2600 (1971) and Buchla Music Easel (1973).

Overview



1. Dry/Wet mix knob
2. Shake slider
3. Tension
4. Springs
5. Treble (Hi)
6. Bass (Lo)
7. Mix CV in
8. Shake in
9. Tension CV in
10. Springs CV in
11. Left and right inputs
12. Left and right outputs

Description

The Spring Reverb module for Modular is based on Softube's stand-alone native plugin and shares a lot of the same features. An addition to the Modular version is the CV control of all parameters. For more detailed information about Spring Reverb, see also its separate chapter.

Parameters

Dry/Wet Mix Knob This knob will set the Dry vs Wet ration for your Spring reverb. For a fully wet signal, turn this knob fully clockwise.

Shake Slider This slider emulates the spring tank being shaken or spring being struck resulting in a thunderous pleasant sound, just like a real spring tank would. When the slider is moved to a position other than halfway between far left and right, the reverb will go "sproing". When the mouse button is released while adjusting this slider, it will automatically revert back to its middle position. Think of it as the slider having a spring-motion (pun intended).

Tension By adjusting this slider the sound of the strings will change from slow and smooth (left) to quick and harsh (right). By setting this parameter differently, many sounds from different reverb tanks can be achieved just by changing the tension and the number of springs (see below). As a general rule, cheaper reverb tanks usually feature high tension and two springs, while a more expensive reverb tanks might feature three springs and a lower tension.

Springs This knob controls the number of springs audible in the Spring-reverb. Fully counter-clockwise, only one spring will make sound, while at 12 o'clock two springs are audible, and finally at fully clockwise the sound of all three springs are mixed to the output.

Treble (Hi) This knob changes the high end equalization of the Spring reverb module output. Tweaking the treble knob clockwise will boost the top-end frequencies above 5 kHz of the wet portion, and tweaking the same knob fully anticlockwise will cut off some of the same top frequencies.

Bass (Lo) This knob changes the low end equalization of the Spring reverb module output. Tweaking the bass knob clockwise will boost the low-end frequencies below 70Hz of the wet portion, while tweaking the same knob fully anticlockwise will cut off some of the same sub frequencies.

Inputs

Mix CV In This CV input controls the Dry/Wet ratio for your Spring reverb and works with the Dry/Wet Mix knob as offset.

Shake In CV input jack that activates the Shake function of the Spring Reverb when it exceeds 0v. Lower CV voltage fluctuations will result in minor "spring strokes", while great leaps (gates etc) will create thundering spring noise-effects. This jack works in tandem with the Shake slider as offset.

Note: automating the shake slider from a performance module can slightly alter its functionality since the control will not have the spring function of the module's own control.

Tension CV In This CV input jack controls the tension of the springs. Swiftly modulating this input with a sine or triangle wave can create desirable chorus-like sounds.

Springs CV In CV input jack with control over the output volume of the three springs. Low CV will output only one spring, while an increasing CV amount (up to 5V) will add the two other springs in the mix output.

Left In This is the left audio input of the Spring Reverb.

Right In This is the right audio input of the Spring Reverb. It is normalized to the Left audio input when it is not patch, which means that only connecting the Left input via a patch cord will enter mono audio through the Spring Reverb.

Ouputs

Left Out The Left audio output of the Spring Reverb module.

Right Out The Right audio output of the Spring Reverb module.

The Spring Reverb module in use

The Spring Reverb module can be used in a variety of ways, some of them not normally associated with spring reverbs.

1. Use fast modulation of the tension input jack in order to get chorus-like sounds out of seemingly ordinary waveforms. Use this great trick to get animation and blurriness into your patches.

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2. Using a the Spring Reverb in a controlled feedback loop can also create interesting sonic textures. Spring Reverb can be a bit sensitive, especially in higher frequencies; a good practise is to have an envelope-follower controlled filter or VCA in the feedback path along with the Spring Reverb. And by using the envelope follower to simultaneously control the tension of the Spring Reverb in the feedback path,

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3. Another great way to customize your Spring Reverb sound is to use one or many bandpass filters as means of equalization of the output signal.

4. CV control of the Dry/Wet mix sequencer from a synchronised source such as a sequencer can also create cool jerky "frozen" beats reminiscent of 90s Big Beat fame.

5. The shake trigger input can also be used creatively by varying the amplitude of the input CV or pulse you're using, for example via velocity.

Credits

Peter Möller – mathematical modeling. **Torkel Svensson** – mechanical analysis. **Oscar Öberg** – modeling and implementation. **Niklas Odelholm** – framework programming and graphic design. **Kristofer Ulfves** – project management, presets, validation, user manual. **Igor Miná** – user manual layout. **Ulf Ekelöf** – 3D rendering and graphics.

46 Summit Audio EQF-100 Full Range Equalizer

Introduction

IT'S ALWAYS DIFFICULT TO DESCRIBE how an equalizer sounds, but everyone we've asked has described the EQF-100 in the almost the same way, with *rich*, *creamy*, and *smooth* being the most common adjectives. And there is definitely some sort of magic in this equalizer.

With a wide bandwidth setting you'll experience an open sound that will breathe life into your tracks, while a narrower bandwidth gives you a no compromise solution for precision tone shaping.

The four full bands of equalization plus the low and high cut gives you plenty of flexibility for any track or style.

Add a tube stage amplification to perfect the rich tone that comes from the passive EQ design and you'll get a warm and fat distortion when the output VU meter hits the red.

Added Functionality

To make an already sexy equalizer even sexier, we decided to add two beautiful VU meters to the unit. We also added the **Output Volume**, so that you easily can level match the equalizer.

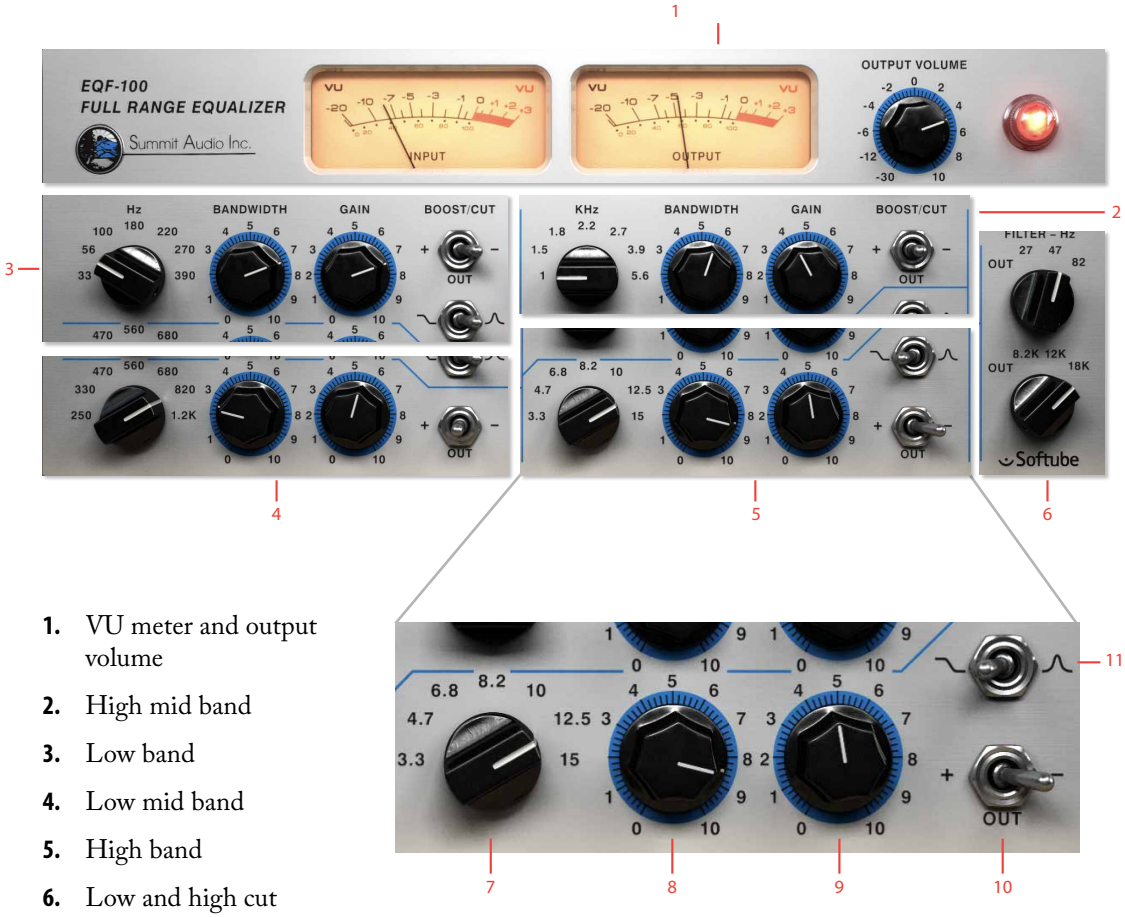
A Note on Distortion

As usual we have included all the distortion that the original unit has, with most of the distortion occuring at the output of the EQ. We've calibrated the distortion so that it will start to saturate when the output VU meter goes up above 0 into the red area. If you do a lot of boosting with the EQ bands and get too much distortion, you can always lower the output volume in order to lower the distortion. The distortion will kick in at around -3 to 0 dBFS.

The sound of the distortion in this unit is very thick and *woody*, as opposed to thin and metallic, which in technical terms means that it contains a strong 2nd order harmonic. For example, the distortion in the TLA-100 compressor has a different character, with more edge than the EQF-100.

User Interface

The user interface consists of four separate bands of equalization (low, low mid, high mid, and high), two cut filters (high and low), the meter section and the **Output Volume**.



1. VU meter and output volume
2. High mid band
3. Low band
4. Low mid band
5. High band
6. Low and high cut
7. Hz, kHz frequency selector
8. Bandwidth
9. Gain
10. Boost/Cut
11. Type Shelving/Bell

Frequency Bands

Each of the four frequency bands consists of seven fixed frequencies with fully adjustable bandwidth (Q) and gain. All bands can be bypassed separately. The lowest and highest bands can also be set to shelving mode.

Hz, kHz Frequency selector Selects one of seven frequencies on which the band boost/attenuate and bandwidth controls operate.

Bandwidth Determines how much of the frequency spectrum is affected by the boost or cut operation. 0 is a narrow bandwidth and 10 is a wide bandwidth.

Gain Determines the amount of boost or cut that takes place.

Boost/Cut This switch sets the section for boost or cut; the center position turns the sections off.

Type Shelving/Bell Changes the high or low band between a shelf or peak type of response curve.

This switch is only available for the highest and lowest bands!

Low and High Cut

Each cut filters consists of three set frequencies and bypass.

Filter - Hz Low Cut Determines the low frequency cut off.

Filter - Hz High Cut Determines the high frequency cut off.

Output Section

Output Volume Adjusts the output volume (before the output distortion) from -30 to +10 dB.

If you want more distortion, just increase the **Output Volume** until the output VU meter hits the red area!

Applications

The following paragraphs are from the original hardware manual, but are of course just as valid for the plug-in as for the hardware. (Well, except for all that stuff about hooking up XLR cables.)

Basic setup

Plug the EQF-100 directly into the insert jack on the mixing console. Put the insert send into the XLR input, and the XLR output back into the insert return of the desired channel. If outboard preamps are being used, try sending the output of the preamp into the EQF-100. This EQ is also perfect before or after a compressor, and for side chain effects such as de-essing.

Boosting a frequency

Select which frequency you want to boost, put the Boost/Cut/Bypass switch to plus and increase the gain. Use the bandwidth control to affect more or less of the surrounding frequencies. A lower number on the bandwidth knob means a narrower bandwidth. If the frequency you want to boost is on the high or low band, set the Shelving/Resonant switch to resonant.

Setting up a low or high filter

Set all four Boost/Cut/Bypass switches to Out. Adjust the two filter settings to the desired frequencies.

Notch filtering

Find the approximate frequency to be attenuated (low, mid-low, mid- high, high). Set the bandwidth to wide (ten), the Boost/Cut/Bypass switch to Cut (minus), and the filter type to resonant (high and low bands only). Turn the gain all the way up (for the highest amount of cut) and start narrowing the bandwidth. Use the frequency select and bandwidth to narrow in on the frequency to be attenuated. Decrease the gain knob (decreasing the amount of cut) until the frequency is cut with the least effect on the surrounding frequencies.

Low or high shelving

Set the low or high EQ sections to shelving filter. Select the frequency where roll-off or boosting is to begin. Select plus or minus on the Boost/Cut/Bypass switch and turn up the gain knob to cut or boost above the high frequency selected or below the selected low frequency.

Mono and Stereo Operation

The Summit Audio EQF-100 plug-in can operate in both stereo and mono modes. In stereo both channels are linked, but processed individually. How the different modes (stereo, mono and mono-to-stereo) are selected depends on your host software.

Credits

Arvid Rosén – modeling, **Niklas Odelholm** – extra features and modeling, **Mattias Danielsson** – testing, **Torsten Gatu** – framework programming, and **Oscar Öberg** – framework programming, **Ulf Ekelöf** – 3D rendering.

47 Summit Audio Grand Channel

The Tone Sculpter

WE KNEW FROM THE OUTSET of our development of the TLA-100A compressor that we wanted to create a beautiful looking, extremely versatile and gorgeous sounding channel strip plug-in which would combine the TLA and the EQ beast known as the EQF-100. But the EQF-100 proved to be a more difficult creature than we had first imagined.

After having modeled plenty of equalizers we felt confident that absolutely nothing could surprise us and that we would keep all deadlines. We couldn't have been more wrong. The passive design of the EQF-100 was intricate, well-designed and just as beautiful as we thought it would be. We just didn't realize how complicated it was.

But after plenty of work developing a new technique for modeling of big passive circuits we feel confident that you will enjoy this monster.

EQF-100 Full Range Equalizer

Four full bands of equalization, two cut filters, two VU meters and a warm output distortion makes this equalizer extremely versatile. It can be used on any type of source, and once you get to know it, it will become your best friend.

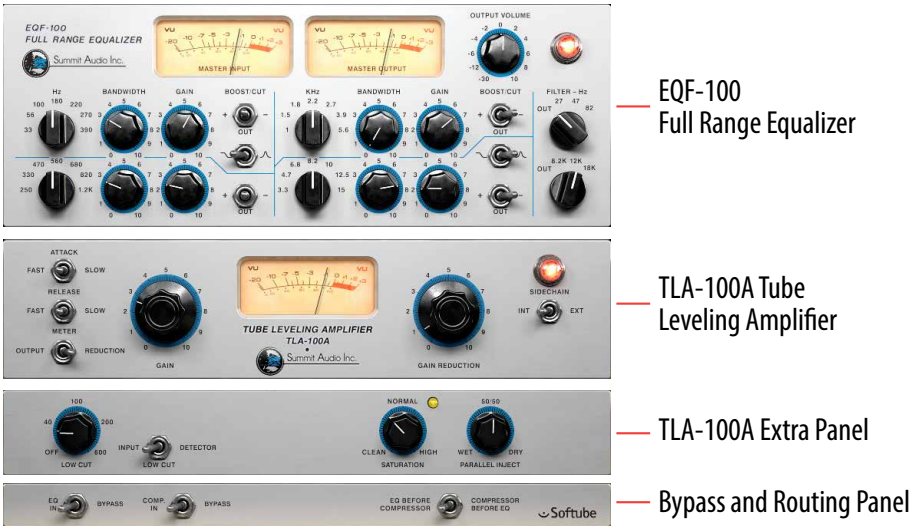
TLA-100A Tube Leveling Amplifier

It doesn't have many knobs, but can create a vast amount of different sounds. With an opto-like compression circuit, you'll get a smooth compression with a very soft knee. But due to its modern and intricate design you can set it to be faster than any other opto-compressor, which makes it more versatile than its competitors.

Different Kinds of Distortion

The TLA-100A has a separate saturation control which lets you adjust the headroom of the unit, so that you can go from a clean sound with almost no audible distortion at all, to a fully saturated leave-no-transients-intact sound. The distortion in the TLA is therefore well-suited for taming transients in drums, but maybe not the first choice when you need to warm up a vocal track.

The distortion of the EQF-100 has a totally different character. Boost the EQ, or crank up the output volume and you'll get a warm and fat type of distortion, due to its strong 2nd order harmonic, that is very well suited for vocals, bass or brass.



EQF-100 Full Range Equalizer

TLA-100A Tube Leveling Amplifier

TLA-100A Extra Panel

Bypass and Routing Panel

User Interface

The user interfaces of the individual units are the same as the individual plug-ins, so please see their respective chapter for more information.

EQF-100 Full Range Equalizer

A fully parametric four band equalizer with two cut filters.

TLA-100A Tube Leveling Amplifier

The classic TLA-100 compressor with adjustable attack and release times and a smooth soft knee character.

TLA-100A Extra Panel

The additional panel adds modern features to a classic compressor. Detector low cut, parallel compression and an adjustable saturation.

Bypass and Routing Panel

The routing panel is used for engaging or bypassing the separate units, and also to select the processing order.

EQ Bypass	Bypasses the EQF-100 equalizer. Right position: BYPASSED. Left position: ENGAGED.
Compressor Bypass	Bypasses the TLA-100A compressor. Right position: BYPASSED. Left position: ENGAGED.

1. EQ bypass
2. Compressor bypass
3. Compressor before EQ

EQ Before Compressor

Compressor Before EQ — Selects if the equalizer should process the sound before the compressor (default) or the other way around. Left position: EQ BEFORE COMPRESSOR. Right position: COMPRESSOR BEFORE EQ.

It is also possible to click on the pilot lamps of each unit to engage or bypass them.

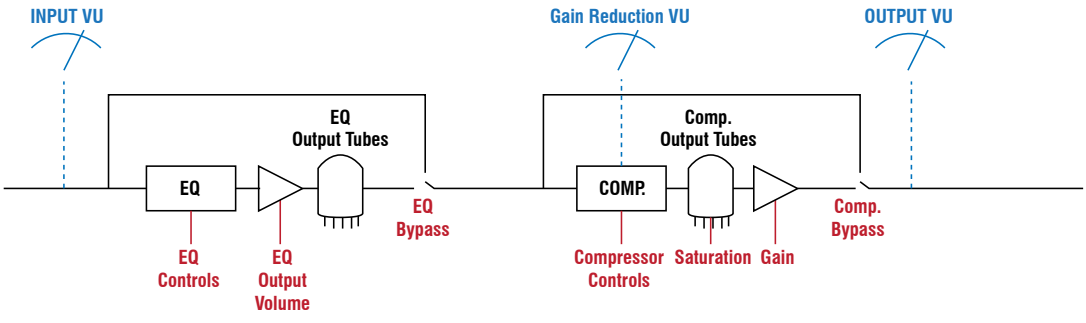
Signal Routing

The signal routing is very straight forward, except for the VU meters on the EQF-100, which are always placed at the master input and the master output. See the block diagram for more information.

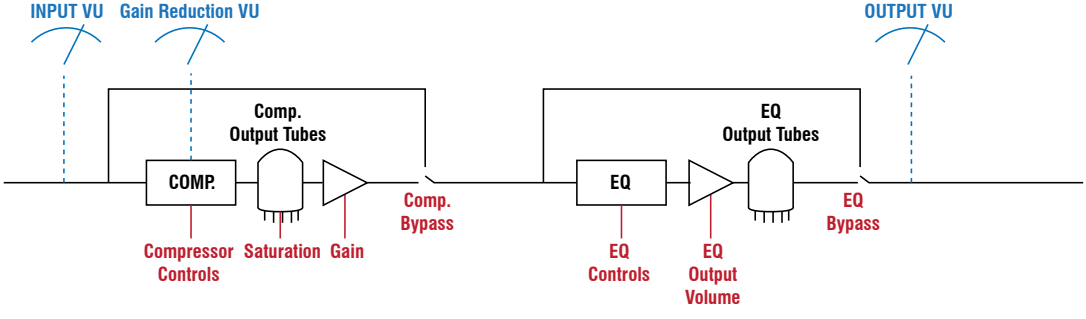
Gain Staging

There are two main volume controls on this unit, the Output Volume on EQF-100 and the Gain knob on the TLA-100A. The best way to use these volume controls is to make sure the apparent audio level stays approximately the same when you bypass the unit. That way it will be easier to bypass and compare the audio, and also to switch order of the compressor and equalizer.

EQ Before Compressor



Compressor Before EQ



VU Meter Calibration

Just like the stand-alone TLA-100A plug-in, the Grand Channel's VU meters are calibrated so that a -9dB$_{RMS}$ signal reads 0 VU.

Sidechain and the Equalizer

No, the equalizer does not affect the external sidechain of the compressor.

Credits

Oscar Öberg – modeling, **Arvid Rosén** – modeling, **Niklas Odelholm** – modeling, **Mattias Danielsson** – testing, **Torsten Gatu** – framework programming, **Ulf Ekelöf** – 3D rendering.

48 Summit Audio TLA-100A Compressor

More Than a TLA-100A

THIS SUMMIT COMPRESSOR IS A CLASSIC, but we felt that there are a couple of features that could really bring this unit into the 21st century. So we decided to add an extra panel below the unit. If you don't care about fancy features, just disregard that panel and the TLA-100A will work just as the original unit.

Low Cut

It is always handy to have the ability to filter out the lowest frequencies so that the compressor doesn't react on them. Here you also have the possibility to choose whether you want to put the low cut on the entire signal (just as if you would have had a Low Cut inserted before the TLA-100 plug-in) or put on just the detector part of the compressor (a.k.a. "sidechain filtering")

Parallel Inject = Parallel Compression

Parallel compression, ie. mixing a compressed signal together with the original signal, is a standard practice today. So instead of using several mix buses to achieve this we added a knob.

Saturation = Headroom = Mix Level Adjust

It's always difficult to choose what level the virtual 0 VU should correspond to in the digital world. Some people use drum loops that peak at 0 dBFS, while others use material recorded at -24 dBFS. That's why we added the Saturation control. With that control you can tune the headroom and the amount of saturation from the tube

output stage in the compressor. Without it, you would have to use the Gain control to set the desired amount of distortion and compensate for that loss or increase in volume with a separate volume control. The Saturation does all that. In one knob.

User Interface

The user interface consists of two panels, the upper panel, containting the original TLA-100A knobs and switches, and the lower panel that offers extra functionality.

1. Attack
   Release
   Meter select
2. Gain
   (output volume)
3. Reduction/Output meter
4. Gain reduction
5. Sidechain

Upper Panel

Attack A three position switch with FAST, MEDIUM and SLOW settings, which corresponds to the time it takes the compressor to respond to the input signal.

Release Controls the time it takes the compressor to return to unity again. A three position switch with FAST, MEDIUM and SLOW settings. The release time is also effected by the program material. The slower the release time setting, the more the program material determines the release time.

Meter Select Allows monitoring the output level (OUTPUT) or the amount of gain reduction taking place (REDUCTION).

Gain Output volume. Unity gain is at "25". The value display in the lower menu bar will show 0.0 dB.

Gain Reduction Sets the amount of gain reduction and the operating point where gain reduction begins. The higher the gain reduction, the higher the ratio becomes. Works almost like a threshold control.

Sidechain If you have an external sidechain present, switch to ext to use the external sidechain as target signal for the compressor. Use when you want a signal to duck under the external signal (for instance ducking strings under a bass drum, or music under a voice over).

Lower Panel

Low Cut Adds a low cut filter on either the detector or the input. From OFF to 600 Hz. OFF disables the low cut functionality.

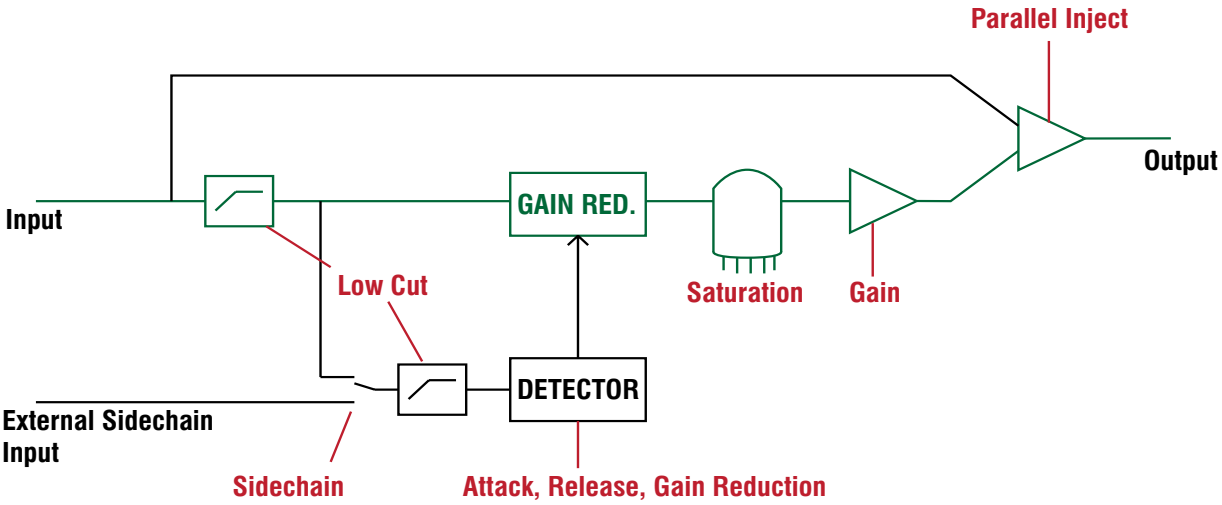
Low Cut Selector Select if the low cut filter should affect the input: like a normal low cut filter placed before the TLA-100A, or just the detector so that low frequencies doesn't trigger the compressor.

Saturation Sets the amount of headroom in the unit. If you need more distortion, turn up this knob. Distortion can sometimes be very useful together with slower attack times when you get loud transients. Use the **Saturation** knob to limit the transients.

Saturation LED Indicates distortion at the output. You will get the best result if the saturation is used sparingly on percussive sounds, like a drum bus. Let it take care of occasional peaks, and do not run it too hot.

Parallel Inject Adds the un-affected dry signal to the compressed (WET) signal. If you are using the compressor to shape the sound instead of compressing the volume (a rather normal use case these days), it is often useful to blend some of the original signal with the compressed track. An overcompressed drum track has few transients, which will make it sound dull and without energy, but if you blend some of the original transients into that track you'll get the energy back while maintaining the overcompressed sound.

1. Low cut (frequency)
2. Low cut selector
3. Saturation saturation LED
4. Parallel inject



Block Diagram

The drawing above illustrates the functionality of the plug-in. Green lines illustrate the main signal path and red labels the parameters.

The **Low Cut** parameters switch between either the detector low cut filter or the input filter.

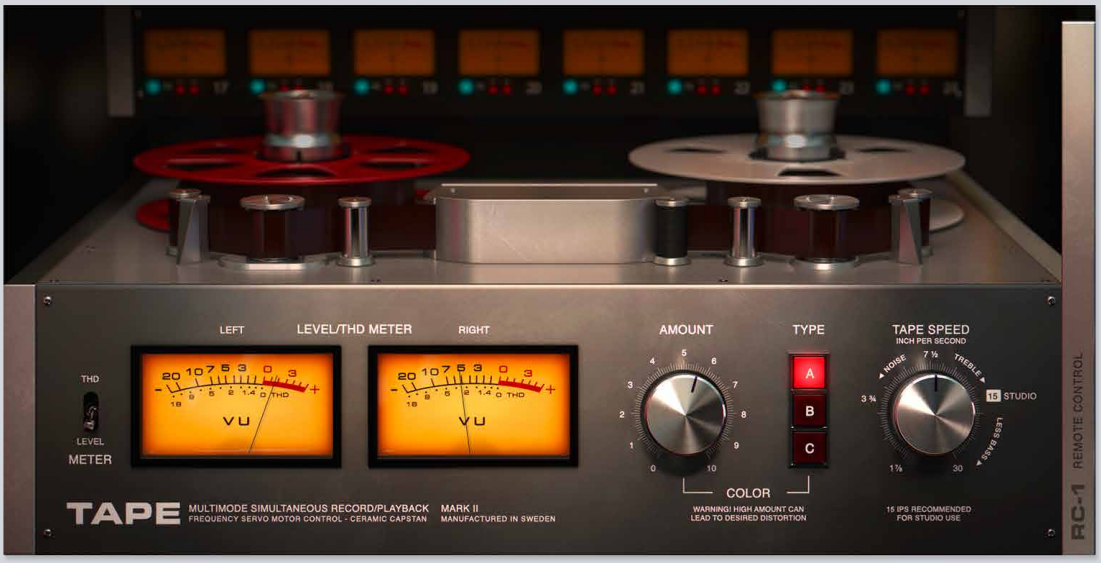
Only one of the low cut filters is active at a time or both bypassed (by setting the **Low Cut** frequency parameter to OFF).

Mono and Stereo Operation

The Summit Audio TLA-100A plug-in can operate in both stereo and mono modes. In stereo mode the plug-in will work as the real unit in "stereo link" mode. This means that both channels will get the same amount of gain reduction (which will preserve your stereo image). How the different modes (stereo, mono and mono-to-stereo) are selected depends on your host software.

Credits

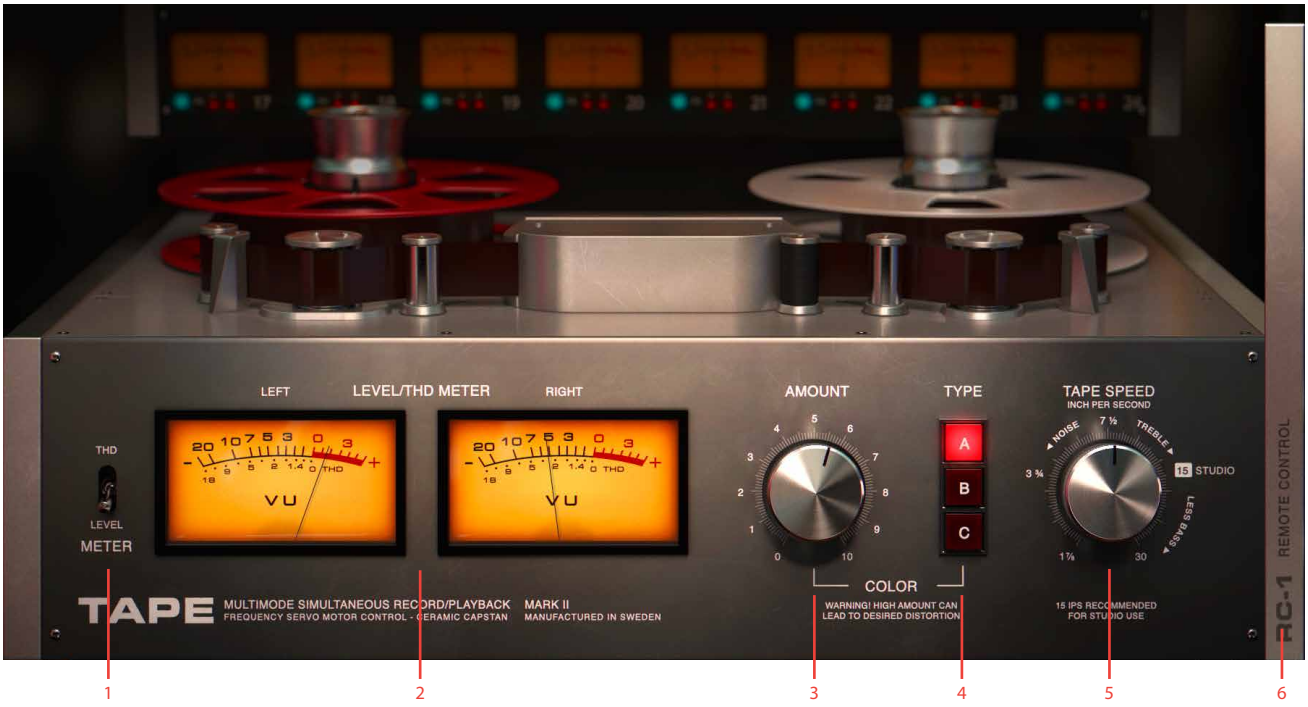
Oscar Öberg – modeling, **Niklas Odelholm** – extra features and testing, **Torsten Gatu** – framework programming, and **Arvid Rosén** – framework programming, **Ulf Ekelöf** – 3D rendering.



49 Tape

AFTER YEARS OF RESEARCH, WE decided to create our own tape emulation plug-in that could easily stand out among other tape emulations. We've modeled the characteristics of the tape machine circuits, tape head, the tape itself and took it even further in a versatile **Tape** plug-in, that can deliver a range of colors from subtle tape vibe to saturated warm tones. Geared with mastering grade **High-Frequency Trim** filter, Tape will shine on master bus and individual tracks as well.

Main View



1. THD/Level meter
2. VU meters
3. Amount
4. Type
5. Tape speed
6. Open side panel

User interface

Clicking the sidebar on the right side opens the panel with additional parameters.

Amount Controls the amount of processing.

Type Selects color types. Each color type is an essence of years of different circuits research combined to a set of iconic flavors delivering instant and rich tape vibe.

Side Panel
with additional controls

1. Dry/Wet
2. Speed stability
3. High frequency trim
4. Crosstalk
5. Noise
6. Input/Output faders and meters
7. Run/Stop

TYPE A is an emulation of Swiss tape machine that gained its popularity in the 60s.

TYPE B tape sound was inspired by a machine with a transformer-based circuit.

TYPE C was inspired by British tape machine with a distinct vintage vibe.

Tape Speed Selects available tape speeds. 30 IPS (inches per second) and 15 IPS are the most neutral speed settings that will deliver mastering grade results and will work great on the master bus. **Speed** settings 7.5 ips and below are extremely useful for creating warm and fat tones.

THD/Level meter Switches the meters between input level meter and THD (Total Harmonic Distortion) meter.

VU Meters Show VU input level and THD amount. When in Level mode, the meters show the VU input level. Keep the input level below the red zone for subtle distortion amount. When in the red zone, the plug-in will add a larger amount of distortion and color.

When in THD mode, the meters show the distortion amount. It is handy switching to this mode to track even the most subtle distortion levels

Peak Meters Show input and output levels of the plug-in. They are useful to match the input and output gains for sound comparison.

Dry/Wet Controls the amount of dry and wet signal.

Speed Stability Controls the amount of tape wear and tear from stable to wobbly.

Using **Speed** stability together with **Dry/Wet** may lead to chorusing sound.

High-Frequency Trim A mastering grade filter that regulates the amount of high-frequency compensation. For the most linear frequency response set the parameter to FLAT. Increasing the amount towards BOOST will add beautiful sheen and clarity to the audio source. For vintage warm tones, move **High Frequency Trim** knob towards CUT.

Crosstalk Is a model of a magnetic tape crosstalk. The parameter controls the amount of crosstalk between channels of a stereo source. When used in Studio One MIX FX mode, it controls the amount of crosstalk between the channels in the session.

See more information about Studio One Mix FX mode at `http://www.presonus.com/products/Studio-One`

Noise Turns on and off tape noise.

Noise levels differ for each **Tape** speed.

Input Input fader controls the input gain. Use it in conjunction with Amount knob to achieve the desired amount of color.

Output Output fader controls the output level of the plug-in.

Run/Stop Run and Stop buttons emulate tape speed up and speed down behavior correspondingly.

In Use

Here are a couple of tips to get started.

Getting Started

tape **Types** are input dependent, so make sure to have correct gain staging, you can check the **Input** or **THD** meters for optimum levels. Insert **Tape** on every track or group of tracks and set the **Amount** until THD meter reads 1 - 1.4. If necessary, increase **Input** gain for more coloration.

Master Bus

To get started, choose 30 or 15 ips and adjust **Amount** to tame excessive transients first, then compensate high end with **HF Trim**. If the mix lacks punch and definition use crosstalk, it will bleed left and right channels in a way magnetic tape does. Use it to reinforce instruments in the mid (vocals, bass and kick). For most mastering applications you'll need up to 40-50% of the crosstalk. If the vocals get too prominent, use **HF Trim** to tame some of the top end.

Use TYPE B if subtle coloration is needed on master bus hence TYPE B has more tolerance than TYPE A and C. You may want to use TYPE C on material lacking punch in the low end.

Electric Guitar Overload

Speed at 3 3/4 and 1 7/8 can deliver a rich harmonic tone to electric guitars. To drive electric guitar, you may need to increase the **Input** gain and **Amount** to the maximum. For warmer guitar tone, use **HF Trim** to tame some of the top end.

Credits

Kim Larsson and **Björn Rödseth** - modeling, **Niklas Odelholm** – graphic design, **Henrik Andersson Vogel** - marketing, **Ulf Ekelöf** – 3D rendering, **Paul Shyrinskykh** - product management and sound design.

50 Tonelux Tilt

Foreword by Paul Wolff

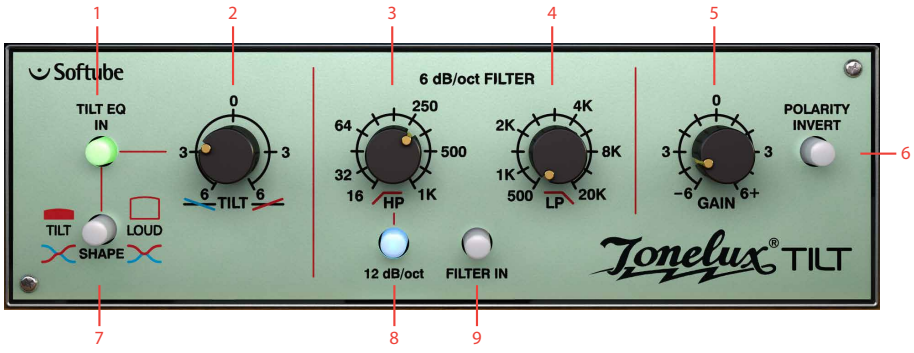
After the success of the TILT control on the MP1 and MP1a Mic Preamps, I felt that a rack mount unit would be a nice addition to the Tonelux product line. The decision was made to make it an 8 channel unit, with polarity, in/out and access with D-subs for ease of use.

After using the TILT rack unit on a few sessions, I was shocked at how useful it was, and how smooth and sweet it made tracks. I tried it on everything from Bass to Over-heads, Guitars, Vocals etc. It did so much with only one knob that the only logical move was to make a plug in. While designing the features, Softube and Tonelux thought that having a few extra options would really fill out the module.

We decided to model the transformer in the Tonelux products, and added a very smooth high and low pass filter, not to fix problems, but to have the abil-ity to use a filter that sounded like it wasn't there. On top of that, we added a loudness feature to the TILT knob, allowing the engineer to boost both low and high at the same time, much like a loud-ness control used in post production, to simulate near and far positioning. So far, the few friends that we have sampled it to have loved it, with comments like "it's just smooth" or "beautiful on over heads, just enough sheen".

After showing the TILT rack unit at a trade show, it was suggested that we consider a live version, which might include some different features, one of which was a way of limiting the boost, but retain-ing the TILT effect, to prevent feedback in wedges and side fills, so Softube came up with a unique "BOOST CEILING" control, where you can set the amount of maximum boost above normal, to prevent feedback, but when using in-ear monitor-ing you can still have the original effect. It works really well to compensate ear fatigue without altering a complex EQ or certain individual chan-nel EQs. It never really gets "louder", it just gets "clearer".

Paul Wolff
Designer and Founder of Tonelux



1. Tilt EQ in
2. Tilt
3. High pass
4. Low pass
5. Output gain
6. Phase invert
7. Shape Tilt/Loud select
8. 12 dB/oct select
9. Filter in

Introduction

THE TONELUX TILT PLUG-IN IS a joint development effort between Tonelux designer Paul Wolff and Softube, bringing the famous Tonelux sound to the digital world. The innovative Tilt design was first featured on the Tonelux MP1a discrete mic preamp module and has been a godsend for engineers and producers that need to take control of their sound in a fast and effective manner. Often, the Tilt knob is all you need to make a track sit better in a mix or to make a track "warmer" or "cooler" and it's perfect in a live situation where you need to tweak the whole mix to adjust for ear fatigue.

The TILT plug-in features not only the original Tilt design but also adds a couple of new features. A special "Live" version of the TILT plug-in is included which is optimized for the DIGIDESIGN VENUE system and contains features requested by live sound engineers. The TILT also includes modeling of the transformer found in the MP1a mic preamp, which adds a subtle distortion for bass frequencies that fattens up the low end of a track.

User Interface

TheTILT is all about getting the sound you want fast and easy. The interface is made up of three sections which will give you flexible and powerful control of your sound.

First section: **Tilt Eq In**, **Shape** and **Tilt**.

Second section: **High Pass**, **Low Pass**, **12 dB/oct** and **Filter In**.

Third section: **Gain** and **Polarity Invert**.

> Want to do a fast mix? Try inserting a TILT on each track of your mix and you'll find that often the **Tilt** knob is the only EQ you'll need.

Tilt Section

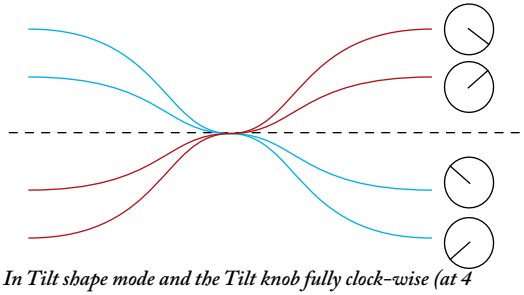
The Tilt section consists of **Tilt Eq In**, which engages the Tilt/Loud equalizer, the **Shape** switch, which lets you select between a Tilt style equalizer or a Loudness equalizer, and finally the **Tilt** knob, with which you adjust the amount of the equalization

When using the **Tilt** knob less is more! Remember that when for example cutting high frequencies, the perception can be that you're boosting low frequencies. Thus you will have double effect when turning the **Tilt** knob.

Tilt Equalization

The **Tilt** control rebalances the whole sound of a track with just one twist of a knob, going from a bright and shimmering top to a warm and wide low-end.

The filter of the **Tilt** is shaped so when one end of the frequency spectrum goes down, the other end goes up, thus cutting frequencies in one end while boosting in the other. The center frequency of this equalizer has been carefully selected to work on a wide range of material and provides a safe way of altering your sound without



In Tilt shape mode and the Tilt knob fully clock-wise (at 4 o'clock), you will get a high frequency boost and a low frequency cut.

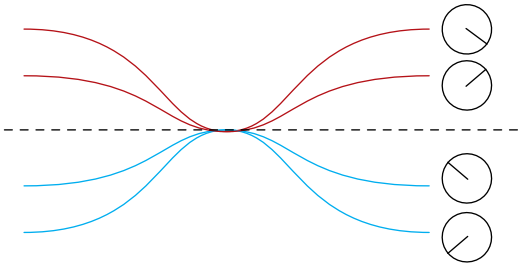
it getting too harsh or too boomy. The gain of the filter goes from 0 to 6 dB.

Loudness Equalization

By pressing the **Shape** button, making it unlit, the function of the **Tilt** knob is changed from TILT to LOUD.

The LOUD setting changes the shape of the filter to that of a loudness control, meaning that when you turn the **Tilt** knob clockwise you will get more bass and more treble and turning it counterclockwise will give you less of both.

Turning the **Loud** control from min to max can give the illusion of a sound going from far away to up close. This is a handy



In Loud shape mode and the Tilt knob fully clock-wise (at 4 o'clock), you will boost both high and low frequencies.

effect for post production.

The **EQ In** button will engage or disengage the TILT/LOUD filter making A/B comparisons easy.

Try using the **Loud** filter with different settings on different sounds that needs to be separated. For example, imagine a guitar track and drum track getting in the way of each other. Boost the **Loud** filter on the drums and cut it on the guitar will place the two in the mix with less clashing frequencies.

Filter Section

The filter section with its two filters will further shape the sound or help you get rid of problematic frequencies in a track. The **Low Pass** filter cut-off frequency goes from 16 Hz to 1 kHz and will attenuate the signal 6dB per octave. When the **12 dB/Oct** switch is engaged the signal will be attenuated 12 dB per octave.

The **High Pass** filter cut-off frequency goes from 20 kHz to 500 Hz and is set to attenuate the signal 6 dB per octave.

By engaging/disengaging the **Filter In** button the **HP** and **LP** filters will be enabled or in bypass respectively.

Output Section

Depending on what version of the TILT you are running, the TILT or TILT LIVE, the knob in the output section will be either a **Gain** control (TILT) or a **Boost Ceiling** control (TILT LIVE). The **Boost Ceiling** is explained below in the TILT LIVE section.

The **Gain** control is a standard make up gain going from -6 dB to +6 dB.

The **Polarity Invert** button inverts the phase when pressed (lit up).

Tilt Live

The TILT LIVE plug-in is optimized for the Avid VENUE system and contains features requested by live sound engineers. The TILT LIVE version does not include the transformer modeling to preserve DSP power and features the **Boost Ceiling** control.

> The TILT LIVE is perfect for in-ear systems when you want to tweak the mix to adjust for ear fatigue, without changing the actual volume of the mix.

In normal operating mode (**Boost Ceiling** on full), the filters in the Tilt/Loud circuit will attenuate some frequencies while boosting others, just like the normal TILT plug-in. For live performance purposes this can be a drawback, since boosting frequencies can sometimes cause feedback in a live monitoring system.

By setting the **Boost Ceiling** on min, you have limited the amount of boost to 0 dB, ie., no boost at all. Whatever setting you put on the TILT LIVE plug-in, it will never boost any frequencies.

> For monitor systems on stage the TILT LIVE can help you changing the balance of a mix without having to risk getting feedback from increasing the volume.

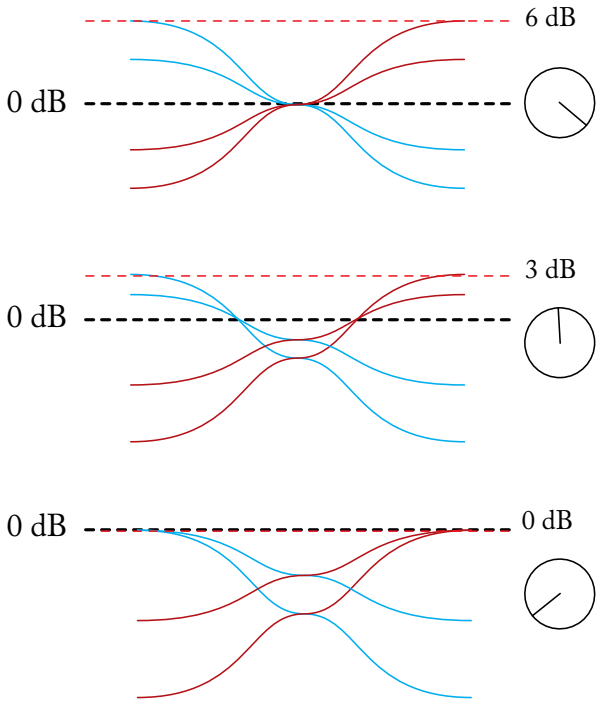
Boost Ceiling

When using the TILT LIVE the **Gain** control will be replaced with a **Boost Ceiling** control.

The **Boost Ceiling** control makes the TILT filter behave differently depending on the **Boost Ceiling** setting.

When the **Boost Ceiling** is at its minimum the audio level will never be amplified. The filters will have the same shape but will always be below 0 dB.

When the **Boost Ceiling** is set to its maximum it will work as the ordinary TILT. When set between, there will be some boosting of frequencies, but never more than you dialed in

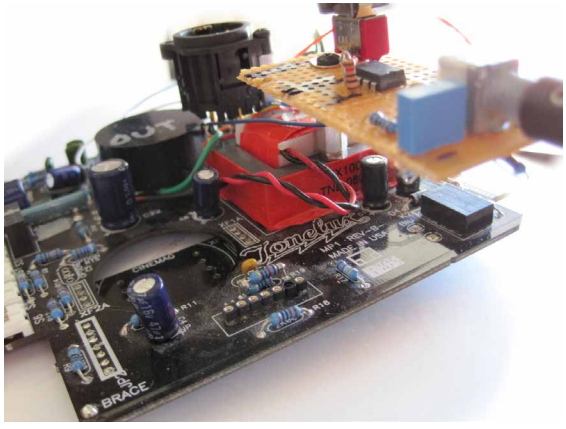


Tilt filter curves for different settings of the Boost Ceiling knob. Top: Boost Ceiling at "In Ear" setting (6 dB), middle: 3 dB, and bottom 0 dB ("Wedge" setting)

Buying Recommendations

The TILT and TILT LIVE plug-ins aren't based on a single piece of hardware. Designer Paul Wolff wanted to create some extra functionality for the plug-in version of his Tilt module, and worked night and day to design a hardware prototype that is the basis of these two plug-ins. However, if you are desperate to find the Tilt control in a hardware package, we can strongly recommend the fantastic sounding Tonelux MP1a mic preamp module

that features the Tilt filters, or the eight channel Tonelux Tilt rack unit.



Here you can see the eight channel Tilt unit by Tonelux, and below is a photo of the prototype Paul Wolff sent us. It looks like, well, a prototype, but sounds fantastic!

Credits

Niklas Odelholm – modeling, **Oscar Öberg** – DSP programming. **Torsten Gatu** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **Paul Wolff** at Tonelux.

51 Transient Shaper

ONE OF THE MOST POPULAR FEATURES of the Console 1 is the addition of the Transient Shaper in the Shape section. The ability to quickly get expanding or compressing effects by a quick tweak of a single knob is extremely useful and in many cases completely eliminates the need for additional dynamics processing. While developing the original Transient Shaper for Console 1 we discovered that the underlying algorithm was a very powerful dynamics tool, and that with a couple of small additions we could unleash the true power of it. And in true Softube spirit we wanted to make a product with few simple controls that become a powerful combination that allows you to completely transform your audio track.

The secret ingredient? A cross-over section for the punch or sustain to affect the high or low frequencies. Try it out yourself, and you'll notice how powerful it is. Add a bit of high frequency sustain to some strings or voice. Remove some low frequency sustain on overheads, or combine the two by increasing the high frequency punch and removing some low sustain on a kick.

Shaping vs. Compressing

So how is the Transient Shaper different from your standard dynamics tools, like a compressor? First, it is level independent. This means that it doesn't *need* a threshold control and will react the same even if you change the gain of your track. Furthermore, it doesn't care about the overall level of a track, it only cares about the transients (**Punch**) or lack of transients (**Sustain**).

While it might be easy to think of the **Sustain** and **Punch** as "threshold" controls on a auto make-up compressor, it is simply wrong to do so. For example, going from the minimum setting on the Sustain up to the maximum, you will start with an effect that is more related to expanding than compressing, and end up with something that sounds similar to a hard "all buttons in" mode compression.

But while it might sound similar to compression and expanding, that's not really what it is, but we won't delve further into those details today.

Knobs and Switches

Here is a brief overview of the knobs and switches of the Transient Shaper.

Sustain Increase or decrease the amount of sustain in the audio. A setting below 0 dB will effectively reduce sustain, for example unwanted ringing in toms or a kick drum. A setting above 0 dB adds more sustain.

Sustain Band Select if the **Sustain** should affect high frequencies, low frequencies or the whole sound (full bandwidth). Increasing high frequency sustain will make a audio source sound more vibrant without cluttering up your mix.

Punch Increase or decrease the transients. Add **Punch** to get more pronounced transients, for example to make it sound as if the drummer is hitting the drums harder. Lower the punch to make the drum hits softer. Punch is only effective on sounds with fast transients, like a drum kit or plucked bass. For "slower" instruments, like voice or piano, it is more effective to work with the Sustain control instead of the punch, although it is possible to soften hard sounds in a vocal track by reducing punch.

1. Sustain
2. Sustain band
3. Gain change
4. Clip LED
5. Output level
6. Punch
7. Punch band
8. Punch type
9. Crossover
10. Output level
11. Clip on output

Punch Band	Select if the Punch should affect high frequencies, low frequencies, or the whole sound (full bandwidth). Increasing high frequency punch is an effective way of adding more "click" to an audio source, such as a kick drum, while decreasing high frequency punch is very useful to tame sharp transients without taking away too much energy from them.
Punch Type	Adjust the timing of the punch. A SLOW punch type is smoother while the FAST punch is more aggressive.
Crossover	The center frequency of the crossover filter.
Level	Output volume
Clip	Turn on or off output distortion. When on, the output soft clips at 0 dB. Clipping in conjunction with increased transients is a very nice way of adding power to each transient without getting too loud output levels. Clipping is indicated by the Clip LED in the meter window. It is also a good idea to set **Clip** to CLIP ON when you're dealing with really loud transients.

Meters

Gain Change	These two meters indicate the gain change in the High and Low frequency bands.
Clip LED	Indicates if the output is clipping
Output Volume	Output peak metering

Slow vs. Fast Punch

There are two main differences between SLOW and FAST **Punch Type**,

1. *Transient detection:* the FAST punch has faster detection, and is better at detecting sharp transients. If the source transient has a slow build-up, it is better to use the SLOW mode, since FAST mode will miss a slow transient.
2. *Transient shaping:* the FAST punch will shape a shorter part of the transient than the SLOW punch, and will therefore sound snappier, as compared to the more fat punch from the SLOW mode.

Typical Use Cases

Here are a couple of typical use cases for the Transient Shaper.

Kick Drum

To get more click in the kick, add some punch in the high band:

Punch: +
Punch Band: HIGH
Punch Type: FAST
Crossover: 700 Hz
Clip: ON

Snare Drum

Decrease low frequency rumble and ringing by decreasing sustain for the low band:

Sustain: -
Sustain Band: LOW

A snare often needs a slower punch to increase the "fatness" of the attack. Let the output clip if the level gets too high, distortion on a snare drum works very good when you want to beef up a drum track:

Punch: +
Punch Band: WIDE
Punch Type: SLOW
Clip: ON

Overheads

Add room and shimmer. Avoid amplifying the low frequency rumble by only increasing the sustain for the high band:

Sustain: +
Sustain Band: HIGH
Crossover: 1-2 KHZ

Drum Kit Enhancement

If you want to shape the entire drum kit, you'll need to try both SLOW and FAST **Punch Type** to see which one is best suited for your recording. Just add a tiny bit of punch for the high frequencies:

Punch: 1-4 DB
Punch Type: SLOW or FAST
Punch Band: HIGH
Crossover: 2-4 KHZ

Vocals

Add shimmer and breath:

Sustain: +
Sustain Band: HIGH
Crossover: 1-2 KHZ

Tame plosives by decreasing the punch for the entire signal. The punch detector will only react on the worst offenders, leaving most of the vocal track intact:

Punch: -
Punch Band: WIDE
Punch Type: SLOW

Bass Guitar

Add body by increasing the low frequency sustain:

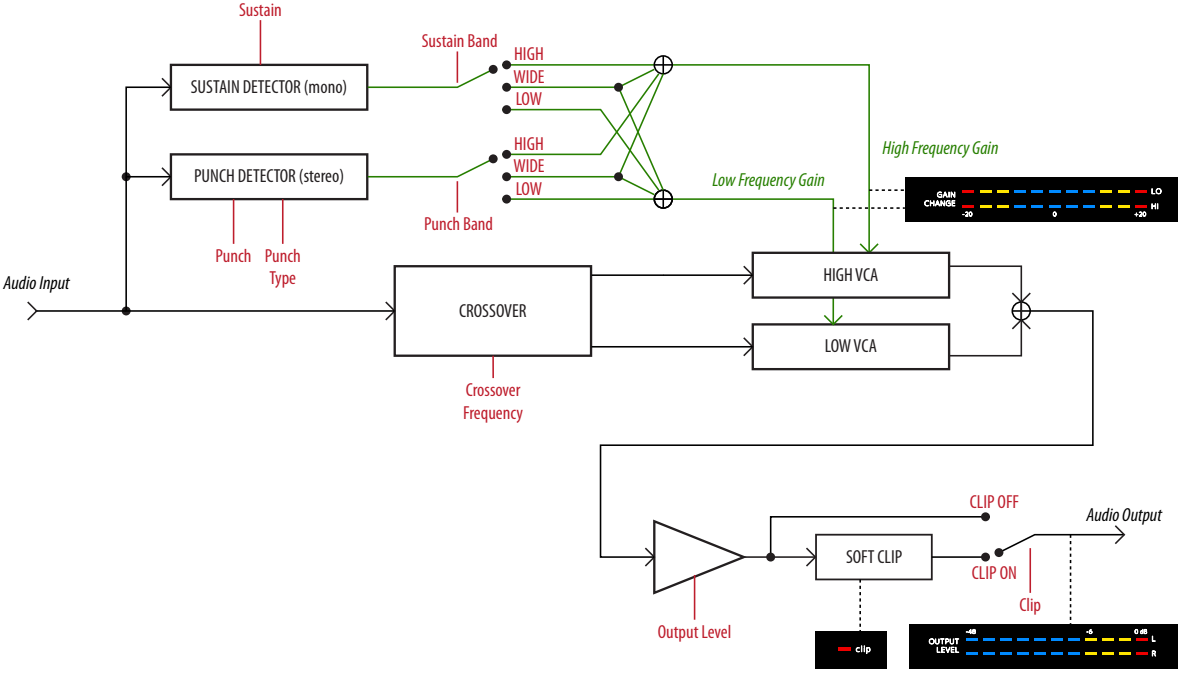
Sustain: +
Sustain Band: LOW
Crossover: 100-200 HZ

Stereo/Mono Operation

The Transient Shaper operates in both stereo and mono. The sustain detector operates on the combined LEFT + RIGHT signal, while the punch detector operates in true stereo.

Block Diagram

The block diagram is a simplification of how it really works, but a useful help to figure out which knob to adjust.



Credits

Niklas Odelholm – concept, modeling, sound design. **Paul Shyrinskykh** – quality assurance. **Patrik Holmström** – framework programming. **Henrik Andersson Vogel** – presets and marketing. **Arvid Rosén** – framework programming. **Oscar Öberg** – framework programming. **Ulf Ekelöf** – graphics rendering. **Torsten Gatu** – framework programming. **Mattias Danielsson** – technical support.

52 Trident A-Range

Foreword by Malcolm Toft

I have evaluated the Softube Trident A-Range equaliser plug-in at my studio with some very experienced recording engineers. After exhaustive listening tests with many different instruments and vocals, I am pleased to say that we all agree this is an incredibly accurate software recreation of my original A-Range design from 1971.

It exhibits all of the nuances of tone from the original, right down to the 'saturation' control which emulates the overdriven artifacts from the original when large amounts of equalisation are applied or the input is driven hard.

Well done to all the guys at Softube.

Professor Malcolm Toft
Original designer of the Trident A-Range console

Torquay, Devon, England
December 17, 2009

Background

THE TRIDENT™ A-RANGE® IS A LEGENDARY piece of equipment. As only thirteen A-Range consoles were ever made, it remains a holy grail for sound engineers and producers around the world that hope to create equal works of art to those that were originally recorded on the A-Range – David Bowie's Ziggy Stardust, Queen, Elton John, the list goes on...

The A-Range was originally designed in the early 70's for Trident Studios in London by Malcolm Toft and Barry Porter. Toft was employed as a sound engineer at Trident Studios and when the studio needed a 24 track recording console, he started Trident Audio Developments to design and build the Trident A-Range. Other products followed and found their way to studios around the world. Toft remains active to this day, developing products under the Trident name.

The A-Range plug-in from Softube is the first and only native plug-in that has been fully endorsed by Toft and Trident Audio.

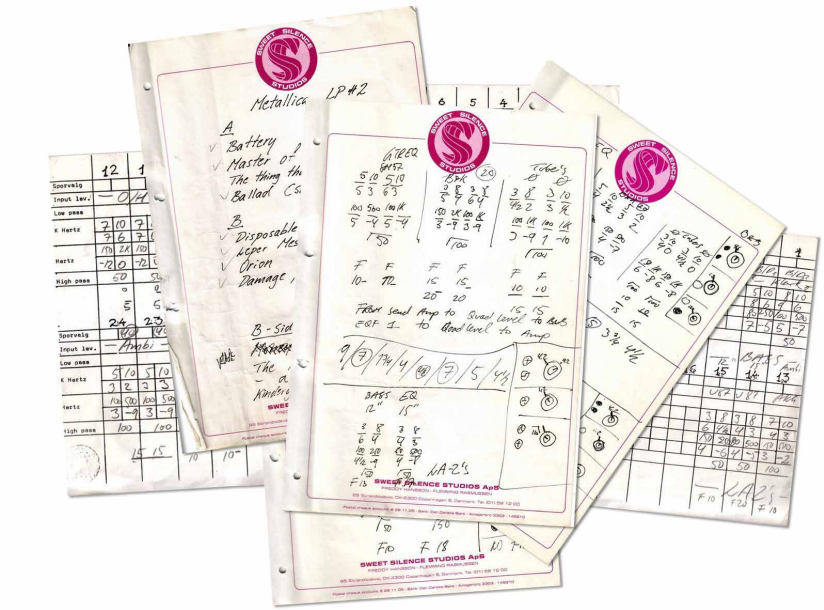
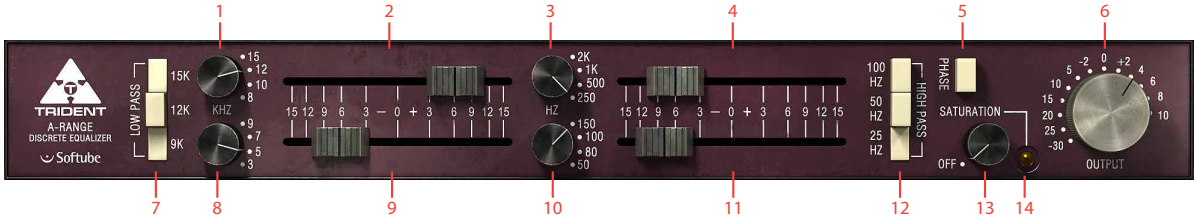
About the Trident A-Range

The plug-in version of the A-Range is modeled after a channel on one of the original 13 Trident A-Range consoles. This particular channel strip is channel number 15 from Sweet Silence Studio 'B' in Denmark and was borrowed from Danish producer Flemming Rasmussen. The A-Range console has been in Rasmussen's care for a long time and used by Flemming Rasmussen when recording albums such as Metallica's *Ride the Lightning*, *Master of Puppets* and *...And Justice for All*.

The A-Range channel features four bands of equalization and high and low pass filters, the sound of the filters are often referred to as "colorful" and "musical". When running a hot enough signal through the original unit and boosting the bands it is possible to get it to distort, the sound is a hairy and effective saturation that is just right in some situation and just wrong in others! Even though the unit was never designed to distort, it has definitely been used this way by renowned producers and we thought it was only appropriate to stay true to the original unit and add the saturation to the plug-in as an option!

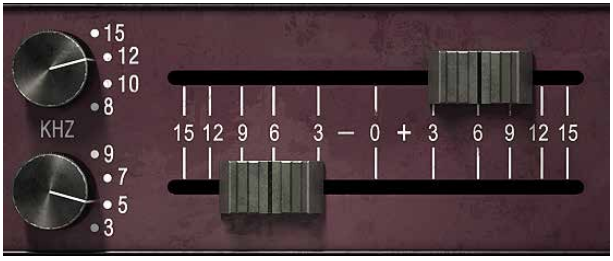
User Interface

The layout of the A-Range equalizer is simple and easy. You have four different bands of equalization, with a frequency selector knob and a gain fader each. **Low** and **High Pass** filters can be activated by pressing the off-white buttons, and a handy **Phase Invert** switch is found on the right side of the panel. The **Saturation** knob engages the console saturation, and is best used as a subtle effect. If you don't wish to get saturation, just put it in the OFF position and all saturation circuits will be de-activated.

The A-Range console was actually designed to have very little distortion, but it was nevertheless possible to crank up the input volume and equalization bands to make it distort, an effect that has been of much use in lots of recordings. The actual amount of saturation depended heavily on the audio material and amplification before the console, so we added the **Saturation** knob to make things easier for you. If all bands are set to zero gain, **Saturation** on its default position (12 o'clock) and **Output Volume** on 0 DB, the A-Range plug-in will have unity gain.



1. High freq.
2. High gain
3. Low mid freq.
4. Low mid gain
5. Phase invert
6. Output volume
7. Low pass
8. High mid freq.
9. High mid gain
10. Low freq.
11. Low gain
12. High pass
13. Saturation
14. Saturation LED

The four equalizer bands have two controls each – a "frequency selector" knob and a gain fader. The fader will boost the signal up to 15 dB by dragging the fader to the right and attenuate the signal by 15 dB by dragging it to the left.



High Shelving Band The frequency options for the **High Shelving Band** are 15 kHz, 12 kHz, 10 kHz and 8 kHz.

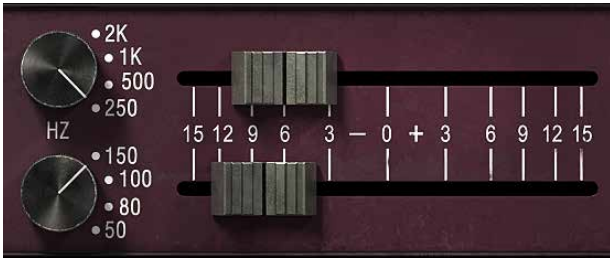
When boosted, the High Shelving Band gives a beautiful high-end shimmer to just about any material. Don't be afraid to try it on a master bus, as the result can be fantastic!

High Mid Band Bell type with the following center frequencies: 9 kHz, 7 kHz, 5 kHz and 3 kHz.

The bell filters interacts with each other in a rather funny way. For example, even if the gain of the Low Mid Band is set to zero, changing the frequency of that band will affect the frequency response of the High Mid Band filter significantly. The results are quite unpredictable and this is part of the reason for the unusual sound of the A-Range!

Low Mid Band Bell type with the following center frequencies: 2 kHz, 1 kHz, 500 Hz and 250 Hz.

Low Shelving Band The frequency options for the **Low Shelving Band** are 150 Hz, 100 Hz, 80 Hz and 50 Hz.



Low Pass and High Pass Filters

The **Low Pass** and **High Pass** filters are controlled by three buttons respectively, which sets the cut-off frequency for the filters – 15 KHz, 12 kHz and 9 kHz for the Low Pass and 100 Hz, 50 Hz and 25 Hz for the High Pass. The cut-off characteristic is unusual and not exactly coherent but the slope is about 12 dB per octave.

The three settings work independently and can be used in any combination which means that the more buttons you engage, the more attenuation you get. For example, pressing all three Low Pass buttons will actually result in a filter with an even lower cut-off frequency than 9 kHz.

Phase Invert

The **Phase** button inverts the phase when pressed.

Saturation

The **Saturation** or distortion of the A-Range is placed after the equalizer bands and is depending on the gain settings of the bands. A louder or "hotter" signal will make the unit distort more and a weaker signal will make it distort less.

To be able to handle signals of different volumes and to make up for equalizer band settings we added a **Saturation** knob that basically is a volume compensated input gain for the saturation circuits. The LED next to the saturation knob indicates when the signal is being distorted. If you want to be sure not to add any saturation to the sound, click the OFF label and the saturation circuit will be bypassed.

You will get the best result if the saturation is used sparingly on percussive sounds, like a drum bus. Let it take care of occasional peaks, and do not run it too hot.

Output Volume

The Output knob controls the output volume from the plug-in. The range of the output volume is -30 dB to 10 dB.

VENUE Stereo Operation

VENUE stereo is not supported due to DSP limit constraints. You will need to use the MULTI-MONO mode instead of STEREO mode. Normal STEREO mode is however available in AAX.

Buying Recommendations

We always make buying recommendations in our manuals for a hardware equivalent of the plug-in. In this case, it's really simple and really hard – If you can find an A-Range channel and can afford it, get it!

Credits

Niklas Odelholm – modeling, **Oscar Öberg** – DSP programming and modeling. **Torsten Gatu** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **Malcolm Toft** and **Barry Porter** at Trident Audio Developments. The original unit is owned by **Flemming Rasmussen** at Sweet Silence Studios.

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This is a photo of channel 15 and 18 from one of the original Trident A consoles. These two are the ones we measured for the A-Range plug-in. A few cosmetic changes have been made to these units while at the Sweet Silence Studios: the original flip switches on the Mute button has been changed to big red switches, the gray fader handles have been changed to bright yellow, and three frequency selector knobs have also been replaced. But it has the original face plate color, a strange but beautiful aubergine-style purple!

53 TSAR-1 Reverb and TSAR-1R Reverb

MODULAR READY

FOR A LONG TIME WE HAD A DREAM about creating the perfect reverb. The most obvious requirement was of course that it should sound extremely good—like a natural room, and better. The reverb tail should be classy and natural, but yet lush and vibrant. It should range all the way from sophisticated halls to a more modulated, sparse and dreamy reverberation. A reverb that surrounds the listener just as a real room does.

While experimenting with different novel reverb designs we realized that the biggest challenge would be to create a product that our users actually could learn how to use. And that led us to the second requirement: it need to be easy to use! We didn't want a reverb with fifty parameters, so complicated that no-one ever dares to change anything, and by that forces the user to use presets.

So once the algorithm was finished we started the work of getting it down to a few, but meaningful, parameters. But don't be fooled by its simplicity, it's one of the most powerful reverbs ever released.

After the original reverb, the TSAR-1, was finished, we decided to go even further and make a natural sounding reverb with just three parameters—and that became the TSAR-1R. It has the same astounding sonic qualities as the TSAR-1 (it's exactly the same algorithm), but focuses solely on natural spaces. And with only three parameters: Time, Predelay and Color, getting a good reverb has never been easier.

Introduction

TSAR-1 STANDS FOR True Stereo Algorithmic Reverb Model 1, and that's just what it is. But TSAR is easier to remember than, well, what ever it stands for...

True Stereo

Both the left and right channels contribute uniquely to the left and right outputs, so the right channel doesn't only affect the right channel but also the left side. Unlike many other reverbs, which might have a two different engines for the left and right sides, or just one engine with different outputs for the left and right sides, a true stereo reverb needs four unique engines to process the audio. This is true stereo, and this is what the TSAR-1 engine does.

Algorithmic Reverb

Even if there are tons of convolution reverbs out there, and anyone with a simple piece of software can create their own set of impulses, there are nothing that compares to a well designed algorithmic reverb.

An impulse response is indeed a very good finger print of a room's acoustics, and does a good job in mimicking a sampled space if you have a completely dry recording. You'll get all the echoes, tails and coloration of that room. An algorithmic reverb is on the other hand designed to add space or ambience to a recording without coloring the dry signal and without interfering with the already recorded ambience. The ability to tailor the reverb to co-exist with the recorded ambience is one of the strengths of an algorithmic reverb, and one very strong reason to why you will always need well designed algorithmic reverbs.

Anatomy of a Reverb

EVERY REVERB DESIGNER has her own way of implementing the user adjustable parameters, but many parameters are the same in most reverb designs and are today just as common in reverbs as bass and treble are in a guitar amp.

Predelay

The Predelay sets the time between the dry signal and the reverb tail. It is often used to achieve the impression of a large room, by making the reverb tail arrive later. A 34 ms predelay corresponds to the time it takes for the sound to travel 10 m. In modern music production, predelay is often used to achieve rhythmic effects, often set at longer times. An 8th notes slap back each in 120 bpm corresponds to a 250 ms predelay.

Reverb Time/Decay Time

This is the time it takes for the volume of the reverb tail to drop 60 dB. A real world analogy for the reverb time could be how much the walls of a room reflect the sound. A big room with much acoustic treatment has shorter reverb time than a stone-wall church of the same size.

Density/Size/Diffusion/Shape

These parameters are used to adjust the sound of the reverb tail. A room with a large size often have longer time between reflections than a small room, and the sound of the reverb becomes less dense. Low density reverbs are very handy if you need a reverb with a long tail that doesn't take up too much energy in a mix. Diffusion alters the build-up of the reverberation, and can have a dramatic effect on percussive sounds.

Early Reflections

The early reflections are the very first echoes that reach the listener and help the listener to decide the size of the room.

User Interface at a Glance

Early Reflections Type The early reflections give the user a sense of the room dimensions.

Early Reflections Mix The mix between the early reflections and reverb tail.

Diffusion High diffusion gives a smoother sound, but also a sound that takes more space in the mix.

Modulation Normal mode is RANDOM , but you can set it to SLOW or FAST for a more chorus-y effect.

Reverb Mix The mix between the direct signal and the reverb signal (including early reflections).

Output Volume Sets the output volume of everything (including dry signal).

Predelay The amount of delay between the early reflections and reverb tail.

Time The time it takes for the reverb tail to decay.

Density Higher density gives a thicker, smoother reverb with more reflections, but also the impression of a smaller room.

Reverb Tone Adjusts the tonality/color of the reverb tail.

High Cut Limits the high frequencies for both the tail and early reflections.

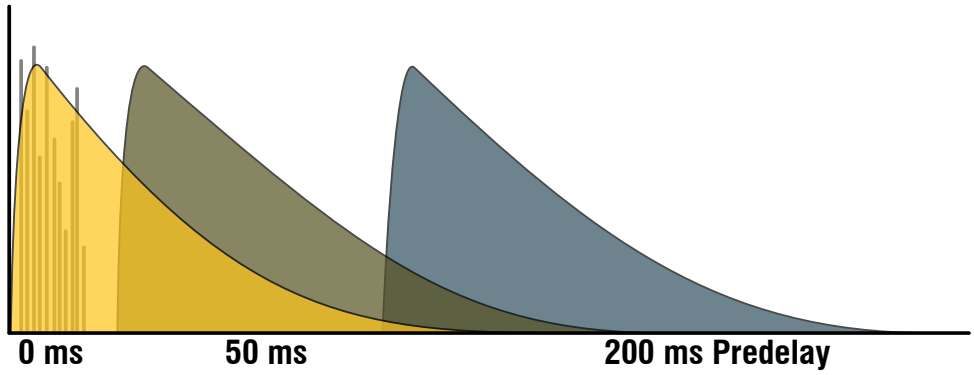
Reverb Parameters

Predelay

Sets the time between the *dry signal* and the reverb *tail*.

The early reflections (gray in the illustration) are **not** affected by the **Predelay** setting.

For a natural sound, there should not be a distinct gap between the initial echos (either from early reflections or from the dry signal) and the **Predelay**. Depending on the the character of the dry signal and the **Early Reflections Type**, the **Predelay** should usually be set below 50 ms.



Predelay and Early Reflections

Since the early reflections are not affected by the **Predelay** and by themselves have an inherent and independent delay, you need to tune the **Predelay** so that it match the length of the early reflections to get a natural sound. For the largest early reflections type, a time of 40–80 ms can be useful. See the **Early Reflections** section for more info.

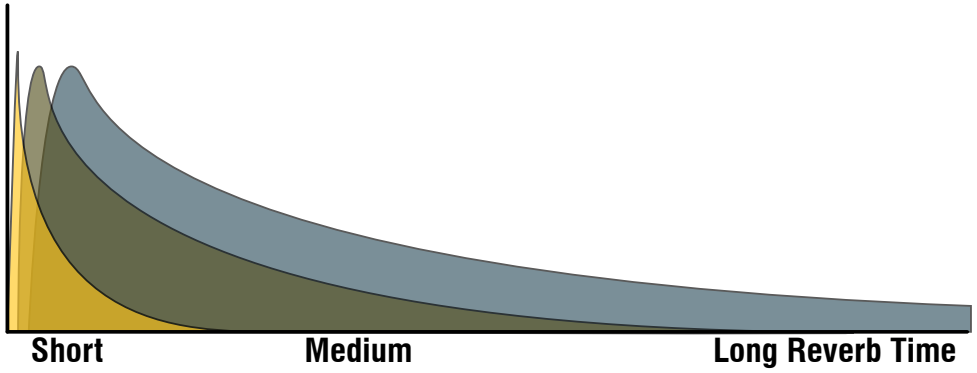
Predelay as an Effect

The **Predelay** is often used to achieve a rhythmic effect, such as a slap back echo on a snare drum, and then with much longer times (about 80–250 ms). A shorter **Reverb Time**, with **Diffusion** and **Density** set to LOW makes this echo more distinct.

Reverb Time

Sets the decay time of the reverb tail.

A longer reverb time gives the impression of a room with more reflective surfaces. If a long **Reverb Time** is used together with a low **Density**, you will get the sound of a large hall. If you on the other hand have a short **Reverb Time** and high **Density**, you will get the sound of a small, tight, studio.



Large Hall Setting

A large hall has often both a long **Reverb Time** and a lower **Density**. Set **Reverb Time** to 1.8–5 s and **Density** to 25–40%. **Early Reflections** on LARGE and about 40 ms **Predelay** enhances this effect.

Room and Chamber Settings

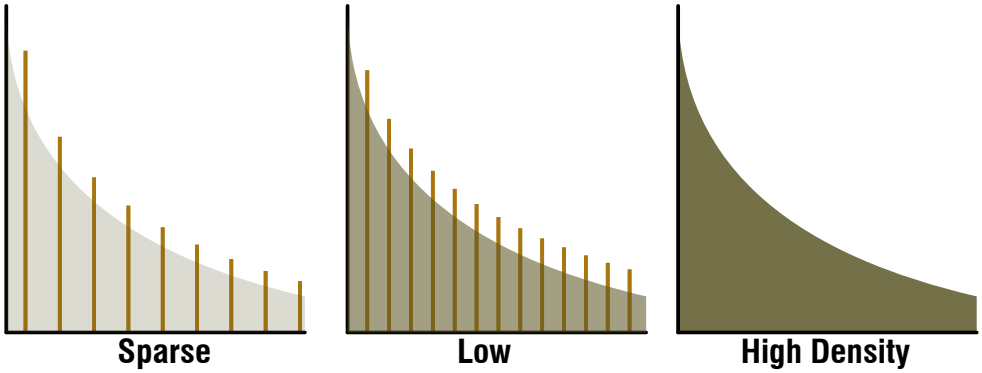
A shorter **Reverb Time** and higher **Densities**, you get the sound of a room or a chamber. A typical room has less than 1 s **Reverb Time** and a **Density** above 50%.

RT60

RT60 is a measurement of the time it takes for the reverb tail to decay 60 dB. The time specified in the display is an approximate RT60 measurement, since the actual RT60 time also depends on the **Tone**, **Density**, **High Cut** and **Diffusion** controls. As usual, use your ears instead of depending on numerical displays.

Density

Sets the density, "thickness", or "smoothness" of the reverb tail. A low **Density** is similar to the sound of a large space, while a high **Density** corresponds to a small space.



Low Density Reverbs

To avoid coloration and muddy-ness in a mix, it is often good to use a less dense reverb. On a non-percussive or legato instrument, such as strings or voice, a sparse reverb will often sound clearer than a dense reverb. Less dense reverbs take up less energy in a mix and is often easier to use, even if they may sound a bit unnatural on its own.

Density vs. Size

Use a high density reverb to get the sound of a small space, and a low density reverb to get the sound of a large space.

Plate and Chamber Settings

To get a sound similar to that of a vintage plate or an echo chamber, you will often need higher density reverbs.

Gated Style Reverbs

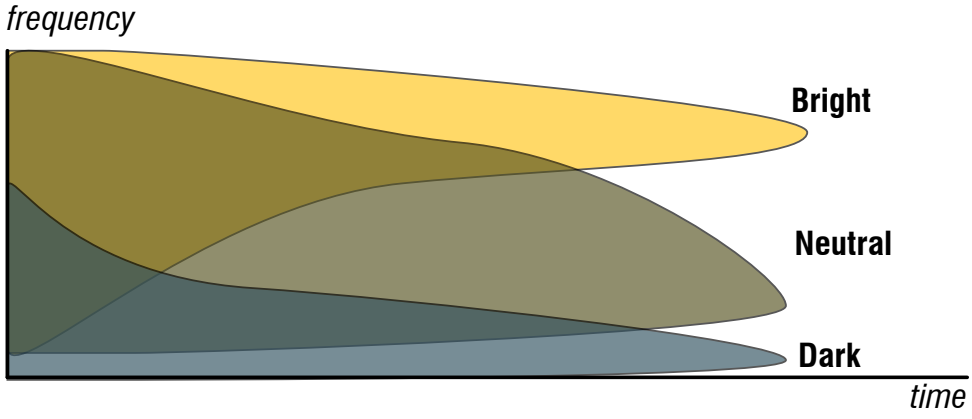
A "gated reverb" is an effect where you gate the reverb signal with the dry signal as side chain to achieve an abrupt cut-off of the reverb tail. You can mimic this sound by setting **Reverb Time** to its shortest value and **Density** below 20%.

Reverb Tone

Use the **Tone** control to adjust which frequencies that have the longest reverb times. With a BRIGHT setting, the high frequencies will decay more slowly, and vice versa with the the DARK setting.

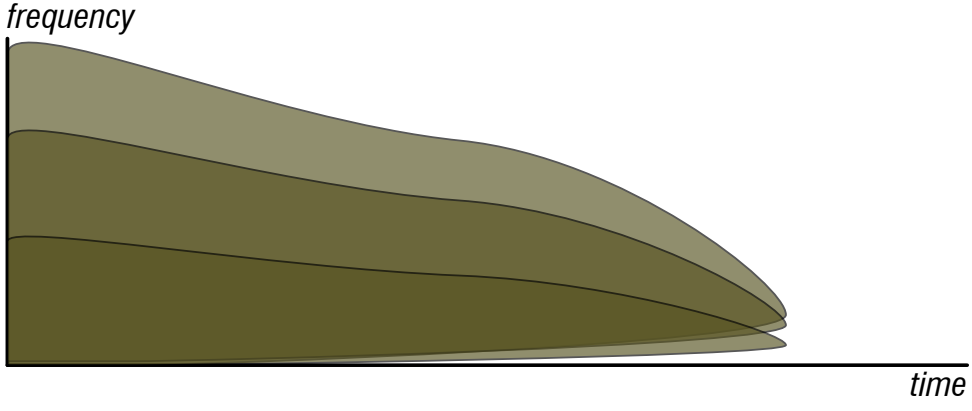
Settings between NEUTRAL and DARK will often give the most natural sounding reverb tails.

The **Reverb Tone** will only affect the reverb tail, not the early reflections.

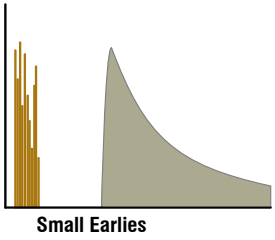


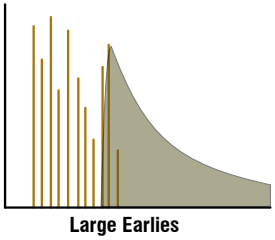
High Cut

Sets high frequency attenuation for both the reverb tail and the early reflections. A cut in the high frequencies often results in a more natural sound.

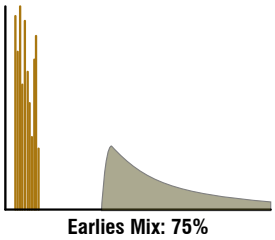


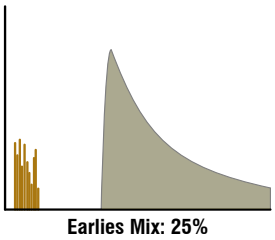
Early Reflections





The early reflections give the listener a perception of the geometry and space. Set the type of geometry with the **ER Type** switch and the blend between late and early reflections with the **ER Mix**. If your material was recorded in a nice room with lots of early reflections, you might not want to impose a new set of early reflections on the recording, and you can thus lower the amount of earlies, or completely turn them off.





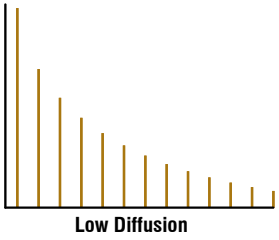
For a close mike recording with very little ambience, early reflections can be very effective in placing the recorded material in a space.

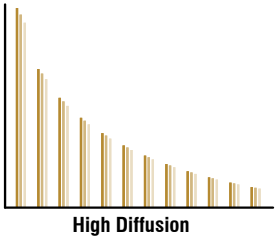
ER Type vs. Predelay

Since the early reflections have built-in delays, it is often necessary to adjust the Predelay according to the **ER Type**. A large **ER Type** may need a longer **Predelay** than a small **ER Type**. The approximate delay times for the three different **ER Types** are listed below:

SMALL: 9–16 ms, MEDIUM: 30–70 ms, LARGE: 45–80 ms

Diffusion





Sets the amount of "smear" in the reverb.

Low diffusion leads to a more immediate sound, sometimes with distinct reflections if the **Density** is low. Set **Diffusion** to HIGH to enhance the sound of percussive sounds.

Although high diffusion often leads to a more natural sounding reverb, it will also make the reverb take up a bigger place in the mix, which sometimes result in a muddier sound. If you, for your application, don't hear much difference between different settings of **Diffusion**, you might as well leave it at LOW.

Modulation

Select between three different types of modulation:

RANDOM: Designed to give a lush and uncolored reverb, without any "pitchy" or "chorus-y" effects. The default setting for most purposes.

SLOW: A more chorus-like modulation with a slow modulation rate. Suitable for extremely long reverb times and legato instruments.

FAST: Same type of modulation as SLOW, but with a faster modulation rate.

All three types of modulation are based on randomization, but RANDOM is just a bit more random than the other two...

Reverb Mix

Sets the mix between the direct and effect signal (reverb + early reflections). When using the TSAR-1 as a send effect, the **Reverb Mix** should always be set at 100% (wet). If you use it as an insert effect, a setting between 15–40% is a good starting point.

Output Volume

The **Output Volume** affects both the direct and effect signal. Usually you can just leave it at 0 dB, but if the output peak meter is going up in the reds it might be a good idea to lower the volume.

Input and Output Meters

The meters on both sides of the display shows the input (left hand side) and output (right hand side) peak levels for both the left and right channel. The meters have a red indicator at 0 dB.

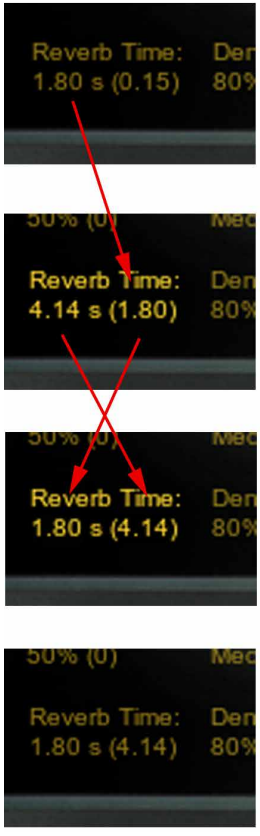
Parameter Display

The display shows the name of the parameter, the value of the parameter, as well as the previous value.

As soon as you start to change a parameter, the display will light up and the former setting will be displayed within the parentheses. While the display is lit up, the former setting will not be over-written.

By clicking on the parameter display, you revert the setting to the previous value.

After one second without changing parameters, the display will get dimmed again.



Presets

The presets are divided into two sections, the *modern* and the *vintage* section. The modern presets focus on application (drums, vocals, orchestral sounds, etc) and the vintage presets mimic sounds from vintage units.

Modern Presets

The modern presets have been designed with a -6 dB reference level, ie., with all faders set at 0 dB the send levels are set at -6 dB. This corresponds to a Mix level of 27% if the plug-in is used as an insert.

Ambience Dark

Opening up the low end is the most transparent way to add space to a source.

Drum Chamber

If you need a little more reverb on the snare or kit than what a room is giving you, but not a full blown plate, the chamber splits the difference and sits down nicely in the mix.

Drum Plate

Sometimes the snare or kit just needs a plate. Here it is.

Drum Room

Dryer recordings can still benefit from a little room sound on the snare or kit. Used in the right

amount, it can make a good recording great, without adding any apparent reverb.

Guitar Hall

A small hall that is the perfect performance space for any acoustic string instrument. Works well with electrics too, when you want to hear the reverb.

Guitar Room

Widen up electric guitar tracks, or provide a little chorused ambience for acoustics.

Hall Dark

If your source has a lot of sharp transients that are jumping out of the track using the other halls, then this is the one to use. This one is very smooth.

Hall Large

This is the place for large orchestral works. The hall is large enough to really let the low end unroll but with a short enough reverb time to retain clarity. age units.

Hall Medium

Brighter and a little less reverb time than the larger hall, this one adds even more clarity to orchestral work. This is the hall to use for vocal ensembles, opera or spoken word.

Jazz Club

Perfect club size space for any performance. Just a hint of slap back to add a live feel.

Percussion Chamber

Highly diffused chamber designed for wood blocks, bells or hand slapped percussion sources.

Scoring Stage

The sound of an A-B mic'd scoring stage. Large and bright, this is the space to use for orchestral production of all sizes for film, or TV.

Studio A

A large studio tracking room big enough for the whole band. Well treated and very smooth, particularly in the high end.

Studio B

A smaller brighter tracking room ideal for the solo performer.

Synth Chapel

Long bright and clear with some sizzle. Huge

size means huge bass as well in this one.

Synth Church

Similar to the chapel, but with a darker decay and more movement in the tail.

Synth Club

Short and full of sizzle. Use this when you want to highlight the source or excite the top end.

Vocal Chamber

If you need a little more reverb on the vocal than what a room is giving you, this chamber is a good alternative.

Vocal Church

A nice long bright church ideal for solo vocals or spoken word. The large predelay helps to retain clarity.

Vocal Plate

Bright and dense. This reverb is meant to be heard.

Vocal Whisper

When used in small amounts, this one brings the sizzle of the performance right up to your ear.

Vintage Presets

In this set of presets we have tried to mimic the sound from old vintage units. We chose to keep the brand names in the presets so that you see where we got the inspiration from.

224 Small Concert Hall

Spacious and expansive—low density initially then builds to a smooth reverb tail.

224 Large Concert Hall

A large empty concert hall with long rolling waves of reverberation.

224 Plate

Smooth and sweet, great on percussion.

224 Room

A medium room/chamber type ambience.

Ambience LRG
Ambience MED
Ambience SM

Carefully tuned and shaped, these add a subtle

presence without sounding too wet.

AMS Ambience

Similar to the "Ambience" preset of the old AMS RMX16 reverb. Great on vocals.

Church
Dark Hall
Dark Stage

These are medium sized spaces with a warmer and more natural ambience than the vintage simulations. Great for spoken word.

Digital Snare
Rock Toms

Similar to a very rare and expensive SONY reverb from the early 1980s. Great for snare and toms.

EMT Hall

Similar to the EMT 250 digital reverb, great on anything.

EMT Plate

Similar to an early EMT mechanical plate reverb. Rich lows and sparkling highs.

Gate
Gate Kick
Gate Snare

Three gated reverb effects, one general purpose, one optimized for kick drum, and one for snare drum.

Plate Small

A tight plate type reverb, very dense and fairly bright.

TSAR-1R Reverb

THE TSAR-1R REVERB USES THE SAME algorithm as its big brother, but is controlled by just three reverb parameters.

Predelay

The **Predelay** has the same function as the TSAR-1 Reverb, but is limited to 200 ms maximum delay time (which is more than enough for natural sounding reverbs).

Time

Instead of just altering the Reverb Time parameter from TSAR-1, this Time parameter adjusts an internal predelay, the early reflections, diffusion, density and decay time. All these parameters have been meticulously fine-tuned to give as natural sounding result as possible for every setting.

Use the **Time** parameter to decide which type of space you want to set your instrument in. Don't be afraid of large halls. If you get too much reverb with a high **Time** setting, use the **Reverb Mix** or the send volume to decrease the amount of reverb.

Color

Set the overall tonal character of the reverb. BRIGHT is useful for strings and vocals, or when you want to give the perception of a highly reflective room, NEUTRAL for normal halls or studios, and DARK for smaller spaces with a lot of acoustic damping.

Automation

The TSAR-1 and TSAR-1R plug-ins fully support automation. But parts of the reverb becomes muted when parameters change in order to suppress noise artifacts ("zipper noise"). Do not use automation to modulate or gradually change parameters. A constantly moving parameter can lead to the reverb being muted.

Mono and Stereo Operation

The TSAR-1 Reverb plug-in is a true stereo reverb and for best performance you should always use it with a stereo output (even if you have a mono input). But it does work in both stereo and mono. How the different modes (stereo, mono and mono-to- stereo) are selected depends on your host software. In most hosts you can select these when you select the plug-in. In other cases, inserting one the plug-in on a mono track will usually make it use the MONO mode, while selecting it on a stereo track makes it use the STEREO mode.

Credits

Oscar Öberg, Niklas Odelholm, Torsten Gatu, and **Arvid Rosén**.

MODULAR READY

54 TSAR-1 and TSAR-1R Reverb for Modular

THE TSAR-1 REVERB IS NOT an emulation. It's not a stock design. It's not a snapshot of a space. The TSAR-1 is a powerful, modern reverb algorithm. It's alive and vibrant, it's gentle and dreamy, and above all – it's natural and believable. The TSAR-1 is the better-sounding alternative to the established, traditional reverb products. At a fraction of the price.

For a full description of TSAR-1 and TSAR-1R, please see its main chapter on page 515.

TSAR-1 User Interface

Both TSAR-1 and TSAR-1R are based on the same reverb algorithm, but uses different controls. TSAR-1 gives the user more power to control the details, while TSAR-1R is easy to dial in, and very easy to get a good natural sound from.

Parameters

ER Type SMALL, MEDIUM, LARGE

The Early Reflections (ER) Type give the user a sense of the room dimensions.

ER Mix 0-100%

The Early Reflections (ER) mix parameter sets the relationship between the early reflections and reverb tail.

Diffusion LOW, MEDIUM, HIGH

This parameter set the smoothness of the sound, but also a sound with high density also takes more space in the mix.

Modulation RANDOM, SLOW, FAST

The modulation parameter sets modulation of the internal feedback loop. For most reverbs Random sounds most natural, but you can set it to Slow or Fast for a more chorus-y effect.

Pre-Delay 0-1000ms

This parameter sets amount of delay between the early reflections and reverb tail.

Reverb Time 0.15 – 15s

This knob sets the time it takes for the volume of the reverb tail to drop 60 dB. In a real world analogy, the reverb time would be how much the walls of a room reflect the sound. For instance, a big room with much acoustic treatment has shorter reverb time than a stone-wall church of the same size.

Density 0-100%

The density parameter sets whether you want a small space with thicker, smoother reverb with more reflections (high density) or a larger, thinner sounding one (low density). Low density reverbs are very handy if you need a long reverb that doesn't take up too much energy in your mix.

Reverb Tone -100 (Dark), through 0 (Neutral) to 100 (Bright)

This knob will adjust the tonality/color of TSAR-1's reverb tail.

High Cut 0.2 – 20kHz

This knob adjusts the cutoff point for the high frequencies for both the tail and early reflections.

Output Volume -inf – 6dB

Sets the output volume of the TSAR-1 module (including dry signal).

Reverb Mix 0-100% (dry to wet)

This knob sets offset mix between the direct signal and the reverb signal (including early reflections).

The TSAR-1 module in Modular builds on the same principles as the TSAR-1 plugin. For more detailed information on the inner workings of the TSAR-1 algorithm, look at the TSAR-1 chapter in the Softube Plug-Ins Manual.

Inputs

Volume CV This input jack let you externally control the overall output Volume of the TSAR-1 module in Modular (including the Dry signal).

Tone CV The Tone CV input jack externally controls the tonality/color of TSAR-1 Module's reverb tail.

High Cut CV The High Cut CV input controls the cutoff frequency of for the tail and early reflections.

In L This is the left input jack of the TSAR-1 module, it is normalized to the right input if no cable is inserted there meaning that a mono signal easily can be spread across the whole reverb.

In R This is the right input jack of the TSAR-1 module.

Reverb Mix CV This CV jack is for external control of the Reverb mix. Incoming CV at this jack is added to the Reverb Mix offset set by the Reverb Mix knob.

Outputs

Out L Main output of the TSAR-1, left channel.

Out R Main output of the TSAR-1, right channel.

Feeding TSAR-1 on to itself can be lots of fun, but it can create unpleasant feedback, be careful not to ruin your ears!

TSAR-1 In Use

Add your TSAR-1 module anywhere in your patch where you want to insert reverb, most commonly would probably be right before patching to main output. Use the **Dry/Wet** knob to adjust the level of reverb in your patch. But you could of course also use the TSAR-1 module as a completely "WET" source and mix it through an external mixer module such as the Audio mixer or X-fade mix modules.

The presets supplied in Modular and Modular FX shows some different uses of TSAR-1 in Modular.

TSAR-1R User Interface

The TSAR-1R is very easy to dial in. Just select the type of room you want (from STUDIO to LARGE HALL) and decide if you want a BRIGHT, NEUTRAL or DARK sounding reverb. That's it!

TSAR-1R Reverb is the little sibling of the more adjustable TSAR-1 Reverb. They share exactly the same high end reverb algorithm, but TSAR-1R offers extreme ease of use for a fast and intuitive workflow.

Parameters

Predelay 0-200ms

The Predelay has the same function as the TSAR-1 Module, but is limited to 200 ms maximum delay time (which is more than enough for natural sounding reverbs).

Reverb Time 0.2 – 15s

In the TSAR-1R module the Reverb Time parameter adjusts an internal pre-delay, the early reflections, diffusion, density and decay time – all at once. All these parameters have been meticulously fine-tuned to give you a natural sounding result.

Color DARK, NEUTRAL, BRIGHT

This parameter set the overall tonal character of the TSAR-1R module. BRIGHT is useful for strings and vocals, or when you want to give the perception of a highly reflective room, NEUTRAL for normal halls or studios, and DARK for smaller spaces with a lot of acoustic damping.

Reverb Mix 0-100%

This knob sets offset mix between the direct signal and the reverb signal.

Output Volume	-inf – 6dB

Sets the output volume of the TSAR-1R module (including the dry signal).

Inputs

Reverb Mix CV	This CV input jack is for external control of the Reverb mix. Incoming CV at this jack is added to the Reverb Mix offset set by the Reverb Mix knob.
Output Volume CV	This input jack let you externally control the overall output Volume of the TSAR-1R module (including the Dry signal).
In L	This is the left input jack of the TSAR-1R module, it is normalized to the right input if no cable is inserted there.
In R	This is the right input jack of the TSAR-1R module.

Outputs

Out L	Main output of the TSAR-1R module, left channel.
Out R	Main output of the TSAR-1R module, right channel.

Feeding TSAR-1 on to itself can be lots of fun, but it can create unpleasant feedback, be careful not to ruin your ears!

TSAR-1R In Use

Add your TSAR-1R module anywhere in your patch where you want to insert reverb, most commonly would probably be right before patching to main output. Use the **Dry/Wet** knob to adjust the level of reverb in your patch. But you could of course also use the TSAR-1R module as a completely "WET" source and mix it through an external mixer module such as the Audio mixer or X-fade mix modules.

The presets supplied in Modular and Modular FX shows some different uses of TSAR-1R in Modular.

55 Tube Delay

Introduction

THE DELAY AUDIO EFFECT HAS A LONG history going back to the times when recording engineers would experiment with analog tape to get a decaying echo effect. Later on, standalone tape delay units were built and even later, delay created with digital technology became possible. Vacuum tubes, or valves, used in audio equipment are known for their unique sonic imprint on sound, something we perceive as "warmth" when the signal passing through it is overdriven.

Tube Delay is a marriage between analog and digital, combining the warmth and natural compression from tubes with the features and versatility of a digital delay. There are three tube sections in Tube Delay, one in the direct signal section, one in the delay feedback loop and one after the feedback loop. The tube sections are created with Softube's patented modeling technology and gives a faithfully recreated sound of a "real" tube preamp.

So, think of Tube Delay as a tube preamp with an added delay section. The tubes in the delay section deteriorates the signal and you can even add more distortion after the delay before blending it with the direct signal passing through the preamp. The effect is perfect for fattening vocal tracks, adding slapback echo to drums or getting a gritty delay for guitars!

1. Delay time
2. Feedback
3. Dry/Wet mix
4. Treble
5. Bass
6. Direct (Dry) drive
7. Delay (Wet) drive
8. Output volume
9. Tempo sync

User Interface

Mix Controls the balance between the delay and the direct signal. Outputs only the original signal through the tube preamp section when set on DRY, and only the delay effect with added tube saturation when set to WET.

Set this control to blend the wanted amount of delay with the dry signal.

Remember! Since there are drive controls for both wet and dry signal paths that affects the volume, you might need to readjust the **Mix** after changing the **Drive** controls.

Direct Drive (top knob) Controls the amount of tube drive in the dry signal path. Because the dry signal path is really a model of a tube preamp, you can use the dry signal path to color or distort audio passing through it like you might use a "real" tube preamp.

Try moving the **Mix** control to full left and notice the coloring of the dry signal path when adjusting **Direct Drive**. This can be useful to just add some warmth to a single track or even a whole mix.

Delay Drive (bottom knob) Controls the amount of tube drive in the wet signal path. Use **Delay Drive** to add color or distortion to the delay signal. This can make the delay signal stand out in a mix and give it a sound that ranges from organic to smashed.

Delay Drive affects the signal after the delay line feedback loop and will thus add distortion to all repetitions of the delay.

Delay Time and Tempo Sync Controls the length of the delay in milliseconds or syncs to host tempo. The total length of delay after the original signal in Tube Delay is one second. In millisecond mode (**Tempo Sync** OFF), **Delay Time** will adjust the time from 1 to 1000 ms. The first half of the control goes from 1 to 100 ms, the second half goes from 101 to 1000 ms. This gives tweaking in the 1 to 100 ms range a greater resolution.

Common **Delay Time** settings are around 40 ms for a doubling effect and roughly 100 to 250 ms for a slapback echo.

When the **Tempo Sync** switch is ON (the "up" position), **Delay Time** is synced to the tempo setting of the host application. The **Delay Time** knob sets the length of the delay in these fractions of a measure: 1/16, 1/12, 1/8, 1/6, 1/4, 1/3 and 1/2. The maximum delay time is still one second though. To get to these values directly, simply click the numbers around the knob. It's also possible to get values in between the fractions by adjusting the knob.

Some hosts don't support reporting Tempo information to the plugin, in these cases a warning will be displayed.

Tempo Sync Here are some examples of the resulting delay time for a given host tempo when the **Tempo Sync** switch is set to ON (upwards position).

	60 BPM	80	100	120	140
1/16	250 ms	188	150	125	107
1/12	333	250	200	167	143
1/8	500	375	300	250	214
1/6	667	500	400	333	286
1/4	1 s	750	600	500	429
1/3	(1 s)	1 s	800	667	571
1/2	(1 s)	(1 s)	(1 s)	1 s	857

1/12 is the equivalent of an eighth note triplet.
1/6 is a quarter note triplet (a common reggae/dub delay rhythm).
1/3 is a half-note triplet.

Feedback Controls the amount of delay repetitions. The higher this setting is, the higher the amount of delay is returned to the input of the delay line.

Feedback goes from no repetitions (just a single delay) to a full feedback with never-ending repetitions.

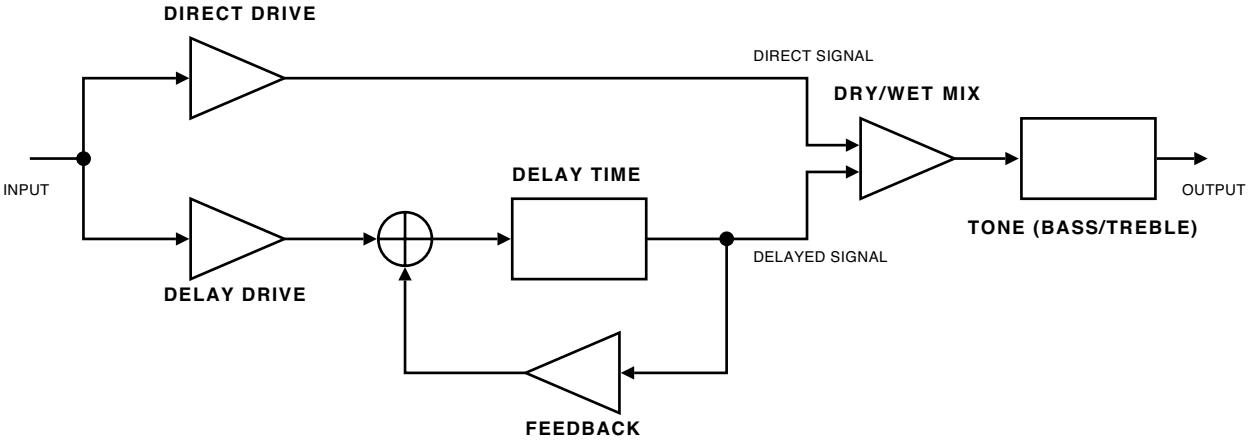
Tone Settings - Bass and Treble Controls equalization of the input of the effect. This affects both dry and wet signal paths and thus also the amount of distortion.

Output Volume Set the level of the output of the plug-in.

Block Diagram

Block diagram of the Tube Delay effect. As you can see, the dry signal (direct signal) isn't very dry, it is affected by both the tone stack and the tube circuits in the **Direct Drive** knob.

For simplicity, the **Tone** control has been depicted as the last stage after the **Mix** knob. In reality, the **Tone** control is embedded within all tube stages, and will thus affect all distortion.



Credits

Torsten Gatu – sound design and framework programming. **Oscar Öberg** – modeling and framework programming. **Niklas Odelholm** – framework programming and graphic design. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering and graphics. **Henrik Midtgaard** – concept.

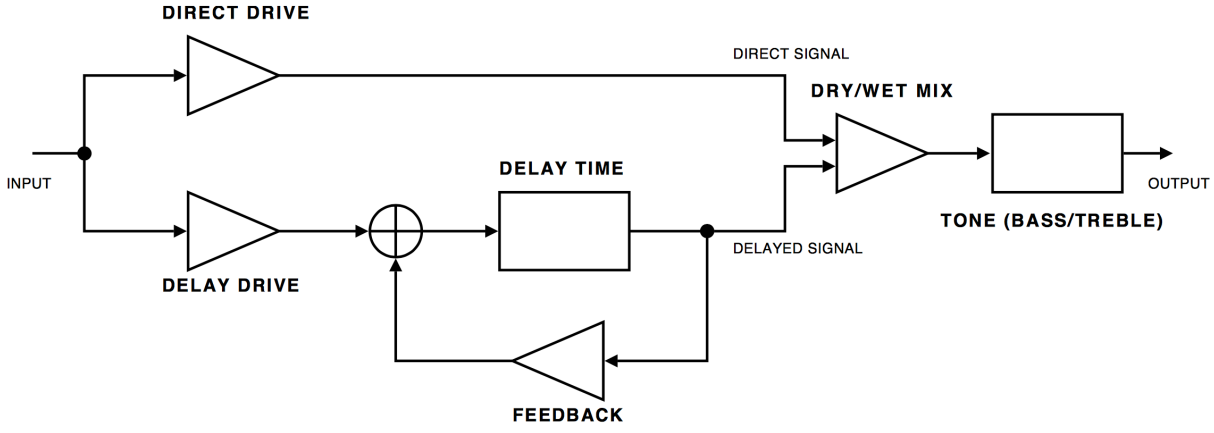
MODULAR READY

56 Tube Delay For Modular

The Tube Delay module for Modular is based on Softube's stand-alone native plugin and shares a lot of the same features. Tube Delay features a dry and a wet path, both with built in tube saturation. The wet path features the delay line which has an analogue tape echo behavior and an additional tube saturation in its feedback path. Added on the Modular module are, of course, CV control capabilities for most of the parameters. Note that the maximum delay time is always one second, which can make Tube Delay unsuitable for certain synced echoes. For more detailed information about Tube Delay, see also its separate chapter.

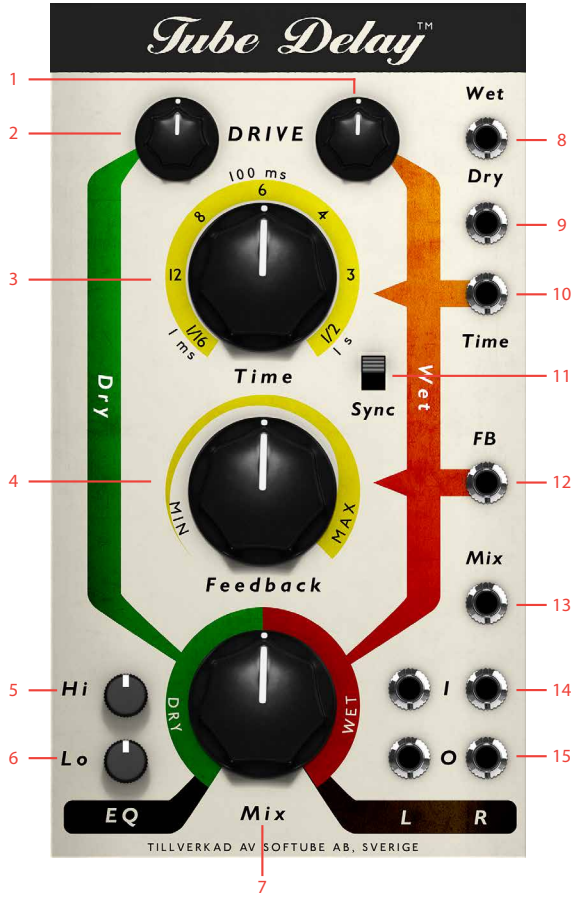
Block Diagram

Below, the a block diagram depicting the Tube Delay is shown. As you can see, the dry signal (direct signal) isn't very dry at all. It is affected by both the tone stack and the tube circuits in the Direct Drive knob.



For simplicity, the Tone control has been depicted as the last stage after the Mix knob. In reality, the Tone control is embedded within all tube stages, and will thus affect all distortion.

Overview



1. Wet drive
2. Direct drive
3. Time knob
4. Feedback knob
5. Treble (Hi)
6. Bass (Lo)
7. Mix knob
8. Wet drive CV input
9. Dry drive CV input
10. Shake in
11. Sync
12. Feedback CV input
13. Mix CV input
14. Left and right inputs
15. Left and right outputs

Parameters

Direct Drive This knob controls the amount of tube drive in the dry signal path.

Delay Drive This knob controls the amount of tube drive in the wet signal path.

Time This knob controls the length of the delay in milliseconds, or, when the Sync switch is set to on, in divisions of a beat. The maximum length of delay after the original signal in the Tube Delay module is one second. In millisecond mode (the sync switch set to off), Delay Time will adjust the time from 1 to 1000 ms. The first half of the control goes from 1 to 100 ms, the second half goes from 101 to 1000 ms. This gives tweaking in the 1 to 100 ms range a greater resolution.

Sync This Tempo sync switch locks the delay time of the Tube Delay to the tempo setting of the host application. The Delay Time knob sets the length of the delay in these fractions of a measure: 1/16, 1/12, 1/8, 1/6, 1/4, 1/3 and 1/2. To get to these values directly, simply click the numbers around the knob. It's also possible to get values in between the fractions by adjusting the knob.

Feedback This knob will affect the amount of the wet signal that is sent back into the wet path. With this knob set around 80% and above (dependent on EQ setting) the delay will go into self-oscillation. Increase or decrease of Treble and Bass will affect this as well.

Mix This controls the mix of Dry and Wet signal in the output.

Treble This knob controls the high end of the built-in equalization in the Tube Delay module. It affects both the Dry and Wet section.

Bass This knob controls the low end of the built-in equalization in the Tube Delay module. It affects both the Dry and Wet section.

Inputs

Wet Drive CV Input A modulation CV (from example a LFO, envelope or sequencer) input here will control the tube-drive of the Wet (i e the delayed) portion of the Tube Delay.

Dry Drive CV Input A modulation CV (from example a LFO, envelope or sequencer) input here will control the tube-drive of the Dry portion of the Tube Delay.

Time CV Input This is a CV modulation input for the delay time. Its range is +-5v.

FB CV Input This is CV control input for the feedback amount.

Mix CV Input This is CV control input for the Dry/Wet Mix amount.

Left Audio Input This is the left audio input of the Tube Delay.

Right Audio Input This is the right audio input of the Tube Delay. The left channel is normalized to the right audio input if no signal is input at the right audio input. This results in mono audio.

Ouputs

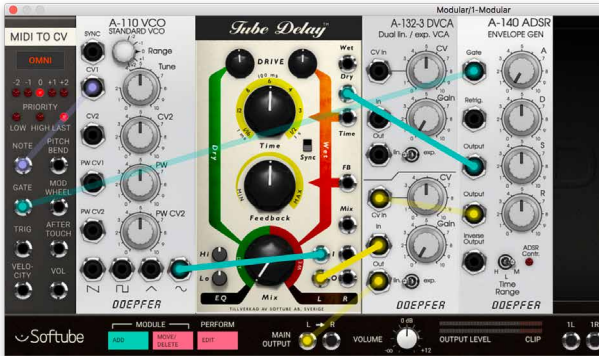
Left Audio Output This is the left audio channel of the output channels of Tube Delay.

Right Audio Output This is the right audio channel of the output channels of Tube Delay.

The Tube Delay module in use

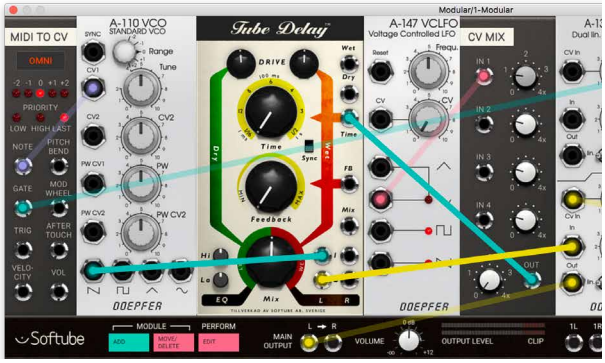
The Tube Delay module in Modular is more than just a delay: because the dry signal path is really a model of a tube preamp, you can also use the Tube Delay module in fully "dry" mode to color or distort audio passing through it like you might use a "real" tube preamp.

1. Use the tube drive of the module for waveshaping of oscillators where the different characters of the dry and wet section can be further enhanced by CV controlling the mix at the same time.



2. Chain the inputs and outputs in a criss-cross pattern to get twice the effect of the drive feature.

3. Subtle and fast modulation of the time parameter, creating variations on a short delay, can successfully be used to create chorus-like effects.



Credits

Torsten Gatu – original sound design and framework **Oscar Öberg** – modeling and framework **Niklas Odelholm** – graphical design and framework **Arvid Rosén** – framework **Peter Möller** – framework **Ulf Ekelöf** – 3D rendering **Kristofer Ulfves** – User Manual, testing and presets **Igor Miná** – user manual layout, photo **Henrik Midtgaard** – original concept.

57 Tube-Tech CL 1B & CL 1B Mk II Compressors

Foreword by John G. Petersen

After several years of consideration, we decided that the time had come to introduce a plugin of our TUBE-TECH CL 1B.

The development of the plugin was initialized in November 2006, as a cooperation between Lydkraft, Softube and TC Electronic. Hearing the result, we found that Softube was able to reach the difficult goal of making a plugin of the CL 1B which came extremely close to the original sound and gave the user all the characteristics of the hardware. After very serious testing of the software, the plugin was released late 2007.

To take this project one major step further, we decided in early spring 2009 to release a VST/AU/RTAS version of the plugin, and once again placed the developing task in the hands of Softube.

We are very impressed by the skills of these young guys from Sweden and we feel excited that the quality of the CL 1B is now available for all workstation users.

We hope you will enjoy the excellence of the TUBE-TECH CL 1B plugin.

Yours sincerely,

John G. Petersen
President, Lydkraft/Tube-Tech

About the CL 1B

The hardware CL 1B differs from many other compressors in that the gainreduction element is made from a nonsemiconductor component, which in itself has a very low harmonic distortion and none of the nonlinearity problems involved when using most semiconductor elements. All parts of these equally important design choices have of course been painstakingly modeled when creating the CL 1B plugin.

Another thing that is special about the CL 1B is the **Attack/Release Select** switch which allows the user to switch between a manual and a fixed attack/release setting, but also makes it possible for the user to combine both the fixed and manual settings. This gives a feature not normally obtained in other compressors: In the combined mode the attack and release controls makes it possible to obtain complex program dependent release-time slopes where a fast peak results in a fast release and vice versa.

1. Gain
2. Ratio
3. Threshold
4. Meter select
5. VU meter
6. Attack time
7. Release time
8. Attack/Release select
9. Sidechain bus select

User Interface

Gain The **Gain** control is used to "make up" for the gain loss, which takes place when the unit is compressing. It is placed after the gainreduction circuit and therefore has no influence on the threshold setting. The **Gain** control is continuously variable from off to +30 dB.

Ratio The **Ratio** control varies the ratio by which the input signal is compressed. If the ratio selected is 2:1, and the input signal increases 10 dB, the output signal is only increased by 5 dB. The **Ratio** control is continuously variable from 2:1 to 10:1.

Threshold The threshold is the point where the compressor begins its action. It is defined as the point where the gain is reduced by 1 dB. The **Threshold** control is continuously variable from +20 dB to –40 dB.

Meter Select Select what the VU meter should display:

INPUT: The meter shows the input level.

COMPRESSION: The VU meter is showing the gain reduction.

OUTPUT: The meter shows the output level.

Please note that it does not show peak or true RMS, it is a VU meter and behaves just like the original unit.

The meter and the plugin is calibrated so that a sine wave showing 0 VU at the output corresponds to a –18 dBFS output signal. Correspondingly, a 18 dBFS sine at the input will show 0 VU if the meter is set at showing the input signal.

Attack Time The **Attack** control chooses how fast/slow the compressor responds to an increase in the input signal. It is continuously variable from 0.5 to 300 milliseconds.

Release Time The **Release** control sets how fast/slow the compressor responds to a decrease in the input signal. It is continuously variable from 0.05 to 10 seconds.

Attack/Release Select This switch selects how the compressor reacts to an increase (attack) or a decrease (release) of the input signal.

There are three settings of this switch:

FIXED: Attack time: 1 millisecond Release time: 50 milliseconds.

MANUAL: Attack time: From 0.5 to 300 milliseconds Release time: From 0.05 seconds to 10 seconds.

FIX./MAN: This setting combines the release times of fixed and manual mode. The attack time is as it is in the fixed mode.

The FIX./MAN. mode always has a fast attack, but it is possible to obtain a release time that depends on the input signal, for example get a fast release when the peak disappears, then superseded shortly thereafter by the release time selected by the **Release** control.

The time the peak disappears to the point where the selected **Release** time takes over, is dependent upon the setting of the **Attack** control. That is, the **Attack** control changes function from a pure attack control to a control of the delayed release with the same time range.

Turn the **Attack** control clockwise to increase the time before the **Release** control takes over. Decrease the **Attack** control to shorten the time before the **Release** control takes over.

This function is valid only if the time of the peak is shorter than the setting of the **Attack** control. If the peak of the program is longer, or if the **Attack** control is set at its minimum position, it will respond just as in the MANUAL mode.

The FIX./MAN. mode acts as an automatic release function with a constant fast attack time and fast release time for short peaks and longer release times for longer peaks. This settings is mainly intended for use on program material (overall compression).

Sidechain Bus Select This knob has two settings:

INTERNAL: Normal mode. The compression depends on the same material as is being compressed.

EXTERNAL: Use an external side chain (if your hosts supports it) to control the gain reduction.

In many plugin formats, such as RTAS, VST3 and AU, it is possible to use an external signal as input to the detector. This is very common if you for instance want to compress a bass line using the bass drum as the trigger to the compressor. In that case, the bass line will be compressed whenever the bass drum is hit.

Monitor the Gain Reduction VU Meter when you set the threshold, since the Input VU Meter will show the original input signal. Just work with the **Threshold** knob until you get enough gain reduction.

> Even if an external side chain signal is present, you will have to activate the external side chain by setting the **Sidechain Bus Select** to EXTERNAL.

Suggested Applications

Here you will find suggestions on various applications of the Tube-Tech CL 1B compressor plugin. They are given as a convenient guide that enables you to familiarize yourself with the different aspects of using the compressor. We have not mentioned specific settings of the **Gain** and **Threshold** as they are dependent on the input levels. Instead we have specified how much compression in dB we feel is needed for the various examples.

These examples were taken from the CL 1B hardware manual, and are of course just as valid for the plug-in as for the real unit.

Overall Compression (Final Mix)

Compression needed: 3-4 dB
Attack/Release Select: FIX./MAN.
Attack: 2 o'clock
Release: 10 o'clock
Ratio: 9 o'clock

Standard Compression (Bass, Piano, Guitar, Keys and Vocals)

Compression needed: 4-5 dB
Attack/Release Select: MANUAL
Attack: 2 o'clock
Release: 10 o'clock
Ratio: 10-2 o'clock

Heavy Compression on Instruments (Line Guitar and Piano)

Compression needed: 10 dB
Attack/Release Select: MANUAL
Attack: 7 o'clock
Release: 1 o'clock
Ratio: 3 o'clock

Compression of Drums (Snare and Bass Drum)

Compression needed: 2-3 dB
Attack/Release Select: FIXED
Attack: –
Release: –
Ratio: 9-12 o'clock

Mono and Stereo Operation

In stereo mode, the gain reduction of the left and right channel is always linked, in order to reduce stereo image shifting. The gain reduction used will be calculated from a combination of the two channels, just as if two hardware CL 1B had been linked together using a sidechain bus.

CL 1B Mk II

Mk II version is enhanced with features not found in the original hardware:

Sidechain Low Cut	Filters the low end from the sidechain signal. Please note that the main signal path is not filtered.
Parallel compression	Sets the amount of parallel processing. For more information on Parallel compression, please check Bob Katz' article on page 623.

Credits

Arvid Rosén – modeling. **Oscar Öberg** – modeling and DSP programming. **Torsten Gatu** – framework and DSP programming. **Niklas Odelholm** – GUI and framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Lydkraft ApS.

Credits – Mk II Version

Paul Shyrinskykh – product management. **Kim Larsson** – DSP and framework programming. **Björn Rödseth, Kim Larsson, Patrik Holmström** – framework programming. **Niklas Odelholm** – graphic design. **Ulf Ekelöf** – 3D rendering. **Daniel Delviken** – marketing. **Maxus Widarson** – quality assurance. **Igor Miná** – user manual layout.

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58 Tube-Tech Classic Channel & Classic Channel Mk II

THE TUBE-TECH CLASSIC CHANNEL contains three different Tube-Tech products, the opto- compressor CL 1B and two Pultec clones; the PE 1C program equalizer (the "classic" Pultec) and the ME 1B mid-range equalizer. On their own, these products immediately and easily produce a beautiful result, and combined they become an extremely powerful tool that lets you shape the sound of a track, bus or the whole mix.

As well as providing these beauties as separate plug-ins, we also chose to combine them into a single powerful plug-in—the Classic Channel. The Classic Channel lets you bypass or engage any of the units, as well as change their order. By bypassing them, they won't take up any of your precious CPU, so you can might as well get used to always use the Classic Channel whenever you need just some CL 1B mojo or Pultec vibe. It makes it easier to add some eq or compression if necessary at a later stage...

Take the time to get to know the individual units and we assure you that you won't get disappointed. There is something about these designs that just make your tracks sound just the way they should... And that's just the reason to why we chose to emulate them.

A Note on Terminology

We have chosen to keep all the labels in the user interface faithful to the original units, even when discrepancies occur. The kilohertz label on the PE 1C is for example labeled "KCS" (kilocycles per second), while it's labeled "kHz" on the ME 1B. Both the peak filters on the ME 1B are labeled as "*High Frequency*" and "*Low Frequency*", while they are referred to in the manual as "*High Mid Frequency*" and "*Low Mid Frequency*" in order to separate them from the real high and low frequency filters in the PE 1C.

User Interface

The user interfaces of the three individual units are the same as the individual plug-ins, so please see their respective chapter for more information.

PE 1C "Pultec" program equalizer

Primary use: Tame the top and low end. Sculpt the overall character of the sound. Use as a high frequency boost.

ME 1B mid-range equalizer

Primary use: More exact sound sculpting than the PE 1C. Vocal and snare drum equalization in the 200Hz—7kHz range.

CL 1B opto compressor

Primary use: It's a compressor...

PE 1C "Pultec" Equalizer

ME 1B Midrange Equalizer

CL 1B Opto Compressor

Bypass and Routing Panel

Bypass and routing panel

The routing panel is used for engaging or bypassing the separate units, and also to select the processing order.

Program EQ In Engages the PE 1C "Pultec" program equalizer. Up position: ENGAGED. Down position: BYPASS.

Mid EQ In Engages the ME 1B mid range equalizer. Up position: ENGAGED.

Compressor In Engages the CL 1B compressor. Up position: ENGAGED.

EQ Before Compressor Compressor Before EQ Selects if the equalizers should process the sound before the compressor (default position) or the other way around. Left position: EQ BEFORE COMPRESSOR. Right position: COMPRESSOR BEFORE EQ.

It is also possible to click the lamps of each unit to engage or bypass them.

1. Program EQ in
2. Compressor in
3. EQ before compressor/Compressor before EQ

Gain Staging

You might get confused by the three different gain controls on the units (PE 1C: **Output Gain**, ME 1B: **Output Gain** and CL 1B: **Gain**). All these volume controls are controlling the output volume of each unit.

If the equalizers are inserted before the compressor, their **Output Gain** controls will affect the **Threshold** and gain reduction of the compressor. If they are inserted after the compressor, they will of course not affect the operation of the compressor.

The best way to use these volume controls is to make sure the apparent audio level stays approximately the same when you bypass the unit. That way it will be easier to bypass and compare the audio, and also to switch order of the compressor and equalizers.

CL 1B Compressor Gain Staging

Just like the stand-alone CL 1B plug-in, the Classic Channel's CL 1B is calibrated so that a -18dBFS signal reads 0 VU.

Sidechain and the Equalizers

No, the equalizers does not affect the external sidechain of the compressor.

Tube-Tech Classic Channel Mk II

MK2 version is enhanced with features not found in the original hardware:

Mid/Side Enables the Mid/Side processing. Please note: when in MS mode, the color of the knobs will change to indicate the current parameter set.

Link Channels Links the parameters of both channels. Please note: when unlinked, it will be possible to equalize and compress LR or MS channels separately. It's recommended to unlink the channels in Mid/Side mode.

Left/Mid Selects the correspondent Left or Middle channel.

Right/Side Selects the correspondent Right or Side channel.

Sidechain Low Cut Filters the low end in the sidechain signal. It's particularly useful in mastering applications to avoid "pumping" caused by low frequencies.

Parallel Compression	Sets the amount of Dry and Wet signal. Please note: the parameter sets the amount of parallel processing for the compressor only, the EQs will not be affected.
Individual Bypass	Each module can be bypassed by clicking on a red lamp. Shift-clicking on any bypass (red lamp) will bypass all 3 modules.

Credits

Niklas Odelholm – modeling, **Arvid Rosén** – modeling. **Torsten Gatu** – framework programming. **Oscar Öberg** – modeling. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Tube-Tech/Lydkraft ApS, Denmark.

Credits – MK II Version

Paul Shyrinskykh – product management. **Kim Larsson** – DSP and framework programming. **Björn Rödseth, Kim Larsson, Patrik Holmström** – framework programming. **Niklas Odelholm** – graphic design. **Ulf Ekelöf** – 3D rendering. **Daniel Delviken** – marketing. **Maxus Widarson** – quality assurance. **Igor Miná** – user manual layout.

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59 Tube-Tech ME 1B Midrange Equalizer

Introduction

"Why do I need another eq?" Well... That is certainly not a new question. But for some reason the Pultec design of equalizers have managed to survive through the decades. The original Pultecs were built in the fifties and Tube-Tech has been making their Pultec clones since the mid eighties. Since the first Pultecs came out there have been tons of inventions that could have rendered this design useless: transistors, op-amps, circuit board designs, fully parametric designs, total recall, and the list goes on. But for some reason, the Pultec character is still desired in pro studios all over the world.

So do you need another eq? There are certainly stuff going on in the Tube-Tech ME 1B that is unique, but its biggest mojo is that it is so well-designed that almost whatever setting you use, it will sound good. And that is good product design, and probably the reason to why everybody needs a Pultec style equalizer.

About the ME 1B

Just like the PE 1C Pultec equalizer, the Tube-Tech ME 1B is a passive, tube-based equalizer. It was designed to accompany the PE 1C equalizer, and provide control over the frequencies that the PE 1C doesn't adjust. On its own it's a great tool to shape the mid- range of any audio material, and together with the PE 1C it becomes a versatile and full featured equalizer.

The focus of the PE 1C, "program equalizer", is the shelving filters and the mid-range boost. It cannot attenuate the mids, and you don't have the ability to get bell shaped filter curves in the lows and highs. The ME 1B solves all these drawbacks by offering the user three bands of equalization: low frequency boost (200–1000Hz), an almost full range sweepable cut (200–7000Hz) and a high frequency boost (1.5-5kHz).

Together they are strong.

User Interface

The user interface of the ME 1B is pretty straight forward. You have three different sections, the **Low Frequency Peak**, **Mid Frequency Dip** and the **High Frequency Peak**. The only confusing thing is that the mid section never boosts, but always attenuate the selected frequency. The other two sections, low and high frequency, boost the frequencies. But as soon as you start to use the ME 1B in conjunction with the PE 1C you'll find that these sections will complement the PE 1C.

In addition to this you get an **Output Gain** control. It isn't included in the original hardware, but is useful in a plug-in.

Low Frequency Section *Mid Frequency Section* *High Frequency Section*

1. Low freq.
2. Low peak
3. Mid freq.
4. Mid dip
5. High freq.
6. High peak
7. Output gain

Low Frequency Section

The low frequency section consists of a sweepable bell shaped peak filter and a gain knob.

Low Frequency Sets the center frequency of the peak filter. The frequency settings are 200 Hz, 300 Hz, 500 Hz, 700 Hz and 1 kHz. All these frequencies overlap with settings in the **Mid Frequency**.

Low Peak The right hand knob adjusts the amount of gain in the peak filter. Adjustable from 0 dB to about 10 dB.

Mid Frequency Section

The mid frequency section consists of a sweepable and attenuating bell shaped filter and a gain knob. With the gain set at 10 you get full attenuation (about -10 dB) from the mid filters.

Mid Frequency Sets the center frequency of the dip filter. The frequency settings are 200, 300, 500, 700, 1000 Hz (overlaps the low frequency section), 1.5, 2, 3, 4, 5 kHz (overlaps the high frequency section) and finally 7 kHz.

Mid Dip Sets the amount of attenuation at the selected frequency. Ranges from 0 dB to about -10 dB.

High Frequency Section

The high frequency section consists of a sweepable bell shaped peak filter and a gain knob.

High Frequency Sets the center frequency of the peak filter. The frequency settings are 1.5, 2, 3, 4, and 5 kHz.

High Peak Sets the amount of gain in the peak filter. Adjustable from 0 dB to about 8 dB. The exact gain depends on the frequency setting.

Output Section

Output Gain The output gain wasn't included in the original ME 1B, but it is pretty useful to have a gain control, especially when you want to A/B between bypassed and engaged mode.

Ranges from -10 dB to +10 dB.

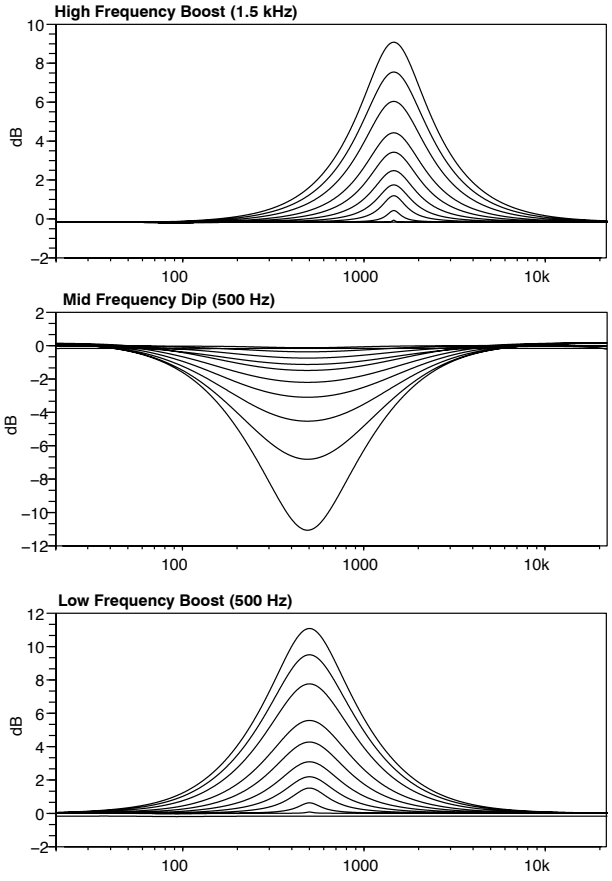
Filter Graphs

Here are some examples of the filter curves for different gain settings.

It is worth to note that the exact shape of these curves depend on three things:

1. What other knobs are set at (they depend on each other).
2. The frequency of the peak or dip filter.
3. The output impedance of the amplifier that is driving (inserted before) the real ME 1B.

In order to satisfy most setups, we chose to model the impedance so that you can get a little bit more gain or attenuation from the filters. That way you can always back it off a bit, if you need the plug-in to match exactly to your hardware setup.



Buying Recommendations

The best way to get as close as possible to the original Pultec Midrange EQ sound in hardware is to get the ME 1B from Tube-Tech. They make great gear. Period. Every time we get to borrow Tube-Tech stuff we immediately fall in love and want to keep it.

Credits

Niklas Odelholm – modeling, **Arvid Rosén** – modeling. **Torsten Gatu** – framework programming. **Oscar Öberg** – framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Tube-Tech/Lydkraft ApS, Denmark.

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60 Tube-Tech PE 1C "Pultec" Equalizer

Foreword by John G. Petersen

In the eighties I got really fascinated by the design and quality of classic tube processors such as Pultec and Teletronics. I felt there was a need to continue that tradition. TUBE-TECH Program Equalizer PE 1A (now PE 1C) was my first unit in production, and it started me off designing the whole TUBE-TECH range. We are excited to experience the PE 1C in the digital domain and impressed once again by the skills of Softube.

Yours sincerely,

John G. Petersen
President, Lydkraft/Tube-Tech

Introduction

"Tube-Tech is Pultec." This is a quote from when the very first Tube-Tech product, the PE 1A, was promoted back in 1985. The vintage US made Pultec EQ 1P (designed and sold during the fifties) was the inspiration for Tube-Tech designer and founder John Petersen who was determined to produce a modern unit that could capture the magic of the original Pultec. The final testing, before shipping the PE 1A, was the EQ 1P and the PE 1A on different channels on a stereo track, making sure that each and every setting behaved exactly the same.

Today, decades later, the PE 1C (with minimal changes from the PE 1A) is still in production and the famous PE 1C has now entered the digital world with the PE 1C plug- in!

About the PE 1C

The Tube-Tech PE 1C is a passive, tube-based equalizer suitable for many different sources ranging from bass drums and guitars to vocals.

The equalizer features a Low Frequency section made up of two low shelf filters which can be combined to

attenuate and boost at the same time. These filters are actually bit apart in frequency even if they are controlled by the same frequency selector. The High Frequency section has a peak and shelving filter working in parallel, which provides a smooth top end while not getting too bright.

The ability to combine the different filters and sections is what gives the PE 1C its characteristic sound and is something which plays a crucial part in explaining the classic Pultec sound.

User Interface

The user interface of the PE 1C, or of a Pultec for that matter, can be confusing if you haven't used a Pultec before. First you have a low frequency section where you select the frequency with one knob, and boost and attenuate frequencies with two different knobs.

The high frequency section is made up of two subsections, the boost and the attenuation section. If you want to boost high frequencies, you select the frequency with the **High Frequency** knob, the width or "Q" of the bell filter with the **Bandwidth** knob, and then the amount of boosting with the **Boost** knob. You'll get a bell shaped boost filter for mid to high frequencies.

But for cutting high frequencies you only get three frequency, choices selectable with the **Atten Sel** knob. The amount of high frequency attenuation is dialed in with the **Atten** knob. This section gives you a high shelving type filter.

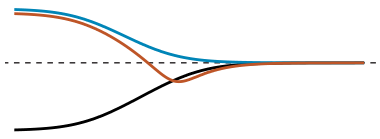
In addition to this you get an **Output Gain** control. It isn't included in the original hardware, but we thought it might be useful in a plug-in.

Low Frequency Section

Boost The **Boost** knob goes from 0 to 14 dB and controls a low shelf filter with a cut-off frequency that is actually a little lower than the ones that are printed on the panel. This gives the unit a unique frequency response when combined the the **Atten** setting.

Atten The **Atten** knob will attenuate the signal 0 to -18 dB using a low shelf filter.

Low Frequency Sets the cut-off frequency for the low frequency **Boost** and **Atten** controls. The frequency settings are 20 Hz, 30 Hz, 60 Hz and 100 Hz.



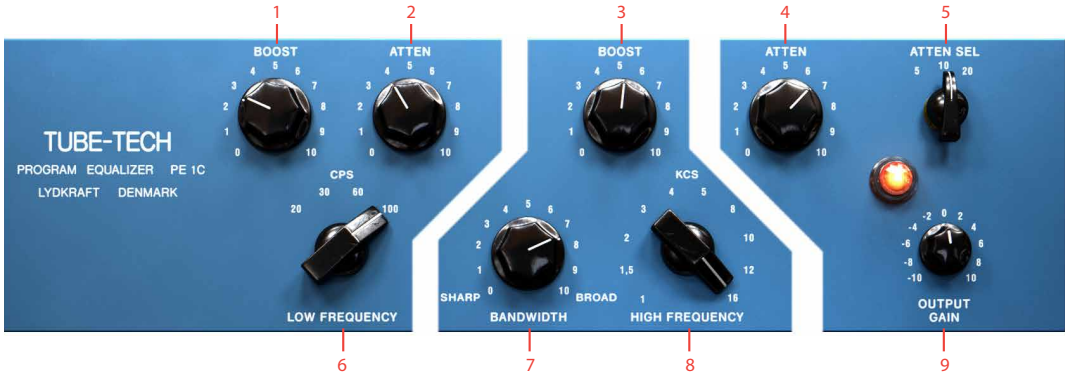
By increasing both the **Boost** (blue curve) and **Atten** (black curve) knobs at the same time, you'll get the very Pultec specific sound (red curve), sometimes referred to as the "*Low End Trick*", which is commonly used for equalizing bass drum. Tube-Tech has some nice instruction videos about this on their web site.

High Frequency Section

Boost Controls the gain for a boost peak filter that goes from 0 to 18 dB for sharp bandwidth and 0 to 10 dB for broad bandwidth.

Bandwidth Sets the width, or Q, of the **Boost** peak filter. Goes from SHARP to BROAD.

Low Frequency Section *High Frequency Section* *High Frequency Attenuation*



1. Boost
2. Atten
3. Boost
4. Atten
5. Atten select frequency
6. Low frequency
7. Bandwidth
8. High frequency
9. Output gain

High Frequency Sets the cut-off frequency for the high frequency **Boost** control. The frequency settings are 1 kHz, 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, 5 kHz, 8 kHz, 10 kHz, 12 kHz and 16 kHz.

High Frequency Attenuation

Atten The Atten knob will attenuate the signal 0 to 18 dB using a high shelf filter.

Atten Sel Sets the cut-off frequency for the high frequency Atten control. The frequency settings are 5 kHz, 10 kHz and 20 kHz.

Output Gain Section

Output Gain The Output Gain knob controls the output volume from the plug-in. The range of the output volume is -10 dB to 10 dB.

Buying Recommendations

The best way to get as close as possible to the original Pultec sound in hardware is to get the PE 1C from Tube-Tech. As mentioned on Tube-Tech's web page, the sound of a PE 1C is exactly like the sound of a brand-new Pultec—not a dusty, old, and unserviced Pultec. The difference is that a meticulously serviced Pultec has a wide and open frequency response, without graininess or harshness. The PE 1C is not a dull and muddy old school equalizer, it's in fact pretty much the opposite!

Anyway, Tube-Tech makes great gear. Period. Every time we get to borrow Tube-Tech stuff we immediately fall in love and want to keep it.

Credits

Arvid Rosén – modeling, **Oscar Öberg** – DSP programming. **Torsten Gatu** – framework programming. **Niklas Odelholm** – DSP programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Tube-Tech/Lydkraft ApS, Denmark.

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61 Tube-Tech Equalizer Collection

Tube-Tech Equalizers Mk II is a combination of the legendary Tube-Tech equalizers PE 1C and ME 1B. We combined them in one unit for better ergonomics and workflow providing perfect tone shaping solution for mixing and mastering. For more information about PE 1C please go to page 556. For more information about ME 1B please go to page 560.

Credits – Mk II Version

Paul Shyrinskykh – product management. **Kim Larsson** – DSP and framework programming. **Björn Rödseth, Kim Larsson, Patrik Holmström** – framework programming. **Niklas Odelholm** – graphic design. **Ulf Ekelöf** – 3D rendering. **Daniel Delviken** – marketing. **Maxus Widarson** – quality assurance. **Igor Miná** – user manual layout.

62 Valley People Dyna-mite

If you haven't used the Dyna-mite before, you will feel extremely confused.

DON'T PANIC!

As soon as you have acquainted yourself with the slightly weird layout and panel labels, you will learn how to master this powerful tool, and soon you will feel comfortable in knowing that you belong to the music industry's version of the "Trekkies", ie., the Valley People fanatics.

Take your time to look through this chapter to get acquainted with the normal modes of operation). During the learning period you will go through these steps:

1. Utter confusion. Desparation. Did I really pay for this?
   Solution: *Read "Basic Limiting", "Basic Expanding" and "User Interface". It's just three pages. Listen to the sound examples on the Softube web site.*
2. Pride. You have learned what the controls do. You think you master every aspect of it.
   Solution: *Read the entire manual, as well as the manual for the original hardware (it's included in the installer).*
3. Awe. You realize all the potential that is in this little thing, and start to think about all the cool things you can do with it.
   Solution: *Do it!*

But don't despair. It doesn't take much time to learn how to use it, it's just that the first 15 minutes can be a bit confusing.

Final word from the developers: *Expanding is the new black!*

About the Valley People Dyna-mite

The Dyna-mite was designed for two main purposes: **limiting** and **expanding**. You choose limiting or expanding by setting the **Mode** switch in either the LIMIT or EXP mode.

Threshold, **Release** and **Output** works as in any limiter or expander. **Range** is a nice addition that enables you to limit the maximum amount of gain reduction.

You can set the type of detector with the **Detector Type** switch (the switch closest to the **Range** knob). AVG is smoother and slower than PEAK. GATE is a mode that works best in the EXP mode.

With the **Det Source** switch you select if you want to use the normal, internal, source (INT) or use an external side-chain (EXT). With an external source you can do many fun things, including **keying** and **ducking**.

You will always get a brief explanation about the current mode in the "Modes Display" beneath the unit.

User interface

1. Threshold
2. Release
3. Range
4. Output
5. Detector source buttons
6. Gain reduction meter
7. Mode buttons
8. Detector type buttons
9. Clip meter

Parameters

Threshold Adjusts the level at which the Dyna-mite starts to expand or limit.

Release Adjusts the time it takes to restore the gain after expansion/limiting.

Detector Source Set it to INT (normal operating mode) as a start. DS-FM and EXT are advanced modes.

Mode Selects main mode. EXP=Expander/Gate, OUT=Bypass, LIMIT=Limiter.

Detector Type Three different ways to detect the signal. Play around and try them out.

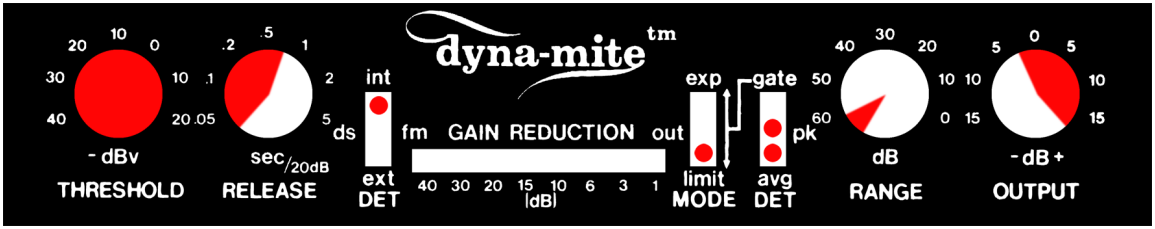
Range Sets maximum amount of gain reduction. Advanced. Leave at 60 dB as a start.

Output Sets output volume. Lower the volume if the output clips to much.

Gain Reduction Meter Displays the current amount of gain reduction.

Clip LED Lights up when the output signal is clipping.

For more detailed explanations of each knob, please see the following chapters.



Basic Limiting

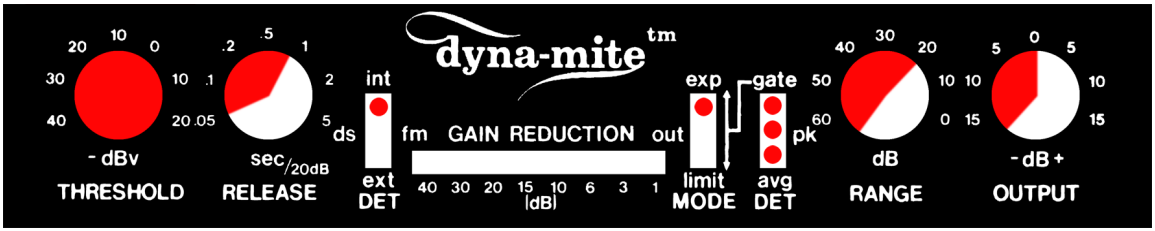
Limiting is a hard Inf:1 ratio compression that can be used to increase the apparent volume, or to even out dynamic differences in a track. It works by reducing the gain for passages that are louder than the selected threshold. The limiting mode on the Dyna-mite is often used to squash drum tracks.

Mode: LIMIT (limiting)
Detector Source: INT (internal source)
Detector Type: AVG (slow attack) or PEAK (fast attack)

Threshold: Adjust to set amount of limiting (as read from the GAIN REDUCTION meter).
Release: Adjust to set release time
Range: 60 dB
Output: 0—15 dB

Procedure

1. Set **Threshold** until you get the desired amount of limiting, as red on the LED array.
2. Set the **Output** volume until you get the desired output volume.
3. Set the **Release** control by ear. If you get too much pumping, increase the release time.
4. Toggle the **Detector Type** between AVG and PEAK to get different attack times.



Basic Expanding

Expanding is when you let the Dyna-mite reduce the gain for passages that are below the select threshold. You can use it as a soft noise gate, use it more aggressively as a hard guitar/drum gate, or you can use to expand the total dynamics of a track.

Mode: EXP (expanding)
Detector Source: INT (internal source)
Detector Type: AVG (slow attack), PEAK (fast attack), or gate (hard noise gate, fast attack)

Threshold: Adjust to set the threshold of expansion
Release: Adjust to set release time
Range: 60—20 dB
Output: –15—0 dB

Procedure

1. Set the **Release** and **Range** to their min positions (CCW).
2. Adjust the **Threshold** so that the desired signals don't light up any LEDs (no gain reduction), while the undesired signals (noise) cause gain reduction (LED array lights up).
3. Set the **Release** control by ear: too fast release time can cause unwanted, abrupt, effects when the signal goes down below the **Threshold**.
4. Set the **Range** to get the desired amount of maximum attenuation.
5. Toggle the **Detector Type** (AVG, PEAK, GATE) to get different kinds of gating effects.

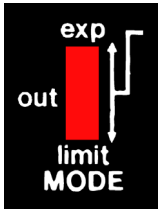
Switches

All three switches work independently of each other, so don't be alarmed. Once you know what each switch does, the Dyna-mite will be easy to operate.

Detector Source (INT, DS-FM, EXT)

Determines the source of the detector.

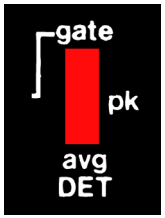
INT	**Internal source**, normal operating mode for **noise gating, expanding** or **limiting**. The gain reduction circuit acts upon the same signal that is fed to the detector.
DS-FM	**Internal source**, but with a detector filter that **boosts the high frequencies**, thus making it easier for high frequencies to trig the detector. This mode can be used for **de-essing**, or it can be used rather creatively when increased sensitivity to high frequencies is desirable.
EXT	**External source**, or "**side-chaining**". The detector is fed a separate signal (a side-chain) and calculates the gain reduction from this signal. This gain reduction is then applied to the main signal. This is used for **keying** (side-chain expanding) and **ducking** (side-chain compression) modes.



Mode (EXP, OUT, LIMIT)

Determines if the Dyna-mite should expand or limit the sound.

EXP	**Expander mode**. Reduce gain for signals **below** Threshold.
OUT	**Bypass**. Output volume knob and output stage clipping still active.
LIMIT	**Limiting mode**. Reduce gain for signals increasing **above** Threshold.



Detector Type (GATE, PEAK, AVG)

This switch determines the ratio and attack times of the detector.

GATE **Fast and aggressive attack**. This is the most extreme mode. Fast attack time (about 50 µs). In LIMIT mode, the ratio is negative (1:–20), which means that for every dB above threshold, the output signal will be reduced by 20 dB! In EXP mode, you'll have a hard noise gate with a 1:20 ratio. So for every dB below the threshold, the signal will be reduced by another 19 dBs.

PEAK **Fast attack time** (about 50 µs), useful for hard limiting of transient material. Inf:1 ratio in LIMIT mode, and 1:2 ratio in EXP modes (for soft noise gating).

AVG **Slow attack.** A slower and more complex attack time (about 1–15 ms). Inf:1 ratio in LIMIT mode, and 1:2 ratio in EXP modes (for soft noise gating).

Detector Type=GATE can be confusing in the beginning, but you can see it this way: In EXP mode, the signal gets gated when the level drops below the threshold (that's normal). In LIMIT mode, the signal gets gated when the level goes above the threshold (that's weird).

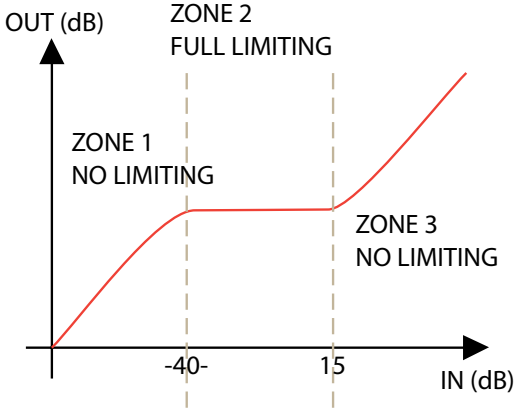
Knobs

Threshold

Classic threshold control. Determines the signal level above which Limiting, or below which Expanding action begins.

Variable from –40 dBv to +20 dBv, which in a plug-in translates to about –60 dBFS to 0 dBFS. A full scale (0 dBFS) sine wave, with **Threshold** set at max (+20 dBv), will cause the Dyna-mite to just start to limit or expand. The exact values depends on the mode selected.

VCA Gain Coupling (a.k.a. Auto Make-Up Gain)
In internal limit and DS-FM modes, the Dyna-mite engages the VCA Gain Coupling which acts like an automatic make-up gain.



Range

This innovative control limits the **maximum gain reduction**. It varies from 0 dB (no gain reduction) to 60 dB **possible** gain reduction.

This control is very useful when expanding, gating or keying. If you use the Keying mode (EXT, EXP, AVG/PEAK) to, for example, duck a background music score whenever a narrator speaks, you can set the **Range** control to 15 dB to prevent the music to be reduced by more than 15 dB.

In the original hardware, the **Range** control was only available in the EXP mode and left out in the LIMIT mode. This was probably due to implementation difficulties, so we decided to introduce the **Range** knob to the other modes as well. (To get the plug-in to behave exactly like the hardware, just set the **Range** knob back to 60 dB in the limit modes.)

Advanced: Range Knob in Limiting Mode (Zones)

When limiting, the **Range** control gives you a very powerful tool to create a "zone" in which the limiting occurs. If you for example limit a drum track and the Gain Reduction meter reads 40 dB in the peaks, you can set the **Range** control to 25 dB and thus get three different zones:

1. No limiting when the drums are below –40 dB
2. Inf:1 limiting in the range –40 dB to –15 dB
3. No limiting for peaks exceeding –15 dB

Since the peaks in the third zone would cause a gain reduction above the limit of 25 dB, and the **Range** is set at 25 dB, no further gain reduction will occur. The effect is that you have a very effective limiting in zone 2, but with the transients of zone 3 being let through. It's different, but sounds somewhat similar to that of a parallel or "*New York style*" compression.

Release

Classic release time control. Determines the rate at which a gain is restored after Limiting or Ducking, as well as the rate at which gain is reduced after Gating, Keying or Expansion attacks.

Variable from 50 ms to 5 sec/20dB.

> **Anticipatory Release Computation**. With fast release time settings and certain material, the gain reduction will follow the signal envelope too tightly, and "pumping" will occur. To escape these effects, yet still allow the user to select rapid release times, Valley International developed a proprietary circuit scheme known as Anticipatory Release Computation (ARC). It analyzes the program input and anticipate conditions which would cause either waveform gain modulation or excessively rapid pumping, and computes a different release envelope in order to prevent, or greatly diminish, these effects. Technical explanations apart, the ARC circuit is definitely a big part of the famous Dyna-mite sound.

Output

Output volume control. Ranges from –15 dBv to +15 dBv. In limiting modes (LIMIT and DS-FM), the output volume will be automatically corrected to fit with the chosen **Threshold** level. In all other modes, the **Output** volume will act as a simple gain control. You will soon find that this auto make-up feature is very handy, since it is extremely easy to make changes to the amount of limiting (with the **Threshold** control) without disturbing the balance of the mix.

Clip Indicator

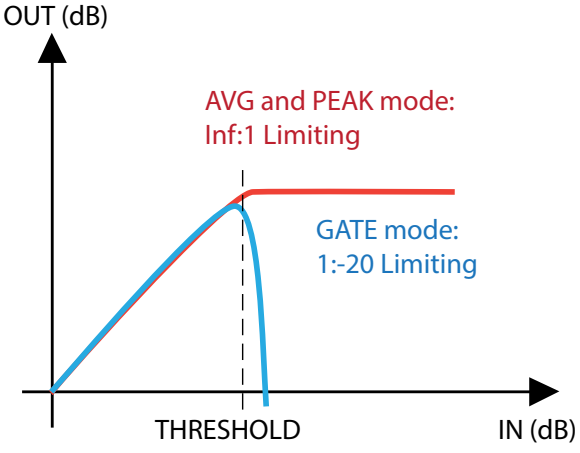
The plug-in features an accurate model of the output stage in the hardware. So whenever the hardware would clip, the plug-in will as well. This is indicated by a clip LED, and just as with the hardware, it is possible for the clip LED to flash, even with very heavy limiting, because the transients will in some cases slip through (for example in the AVG mode).

The original manual states: "*The clipping signal is [often] of exceedingly short duration, and is, in all probability, fully inaudible. Any decision to lower the output level because of sporadic flashing of the clip LED should be based upon what you hear. If the signal sounds "clean", you are probably better off to leave the output level alone*."

Times change, and today we often strive for a "dirtier" sound with distortion and clipping, so feel free to crank up the **Output** volume and experience the brutal distortion of the original Dyna-mite hardware...

Limiting Modes

There are two basic limiting modes: the AVG and PEAK detection. On top of this, you can use the built-in pre-emphasis high frequency filter (**Detector Source** = DS-FM) together with the internal signal. You can also use an external side chain to trigger the limiter (**Detector Source** = EXP), which is usually called **ducking**. Finally, you can set the **Detector Type** to GATE, which will give you **negative limiting**.

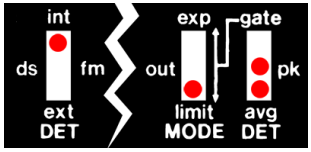


In theory, the difference between AVG and PEAK lies basically in the way the detector calculates the envelope of the signal. AVG uses *Linear Integration Detection*, a proprietary method that Valley International developed, while PEAK uses a more traditional "log-of-the-absolute-value" method.

In practice, the differences lies in the timing. AVG is a bit slower, has a more natural sound, but can be tricky to work with. PEAK is faster and behaves more like a traditional limiter.

You can always toggle between the PEAK and AVG modes and listen to the difference.

Classic Limiting



Mode: LIMIT
Detector Type: AVG/PEAK
Detector Source: INT

Threshold: Adjust to material
Release: Usually short, 0.05-0.5 sec
Range: Typically 60 dB (see section about "Zones" for usage)
Output: Adjust to material, typically above 0 dB.

These are the classic limiting modes. If the detector is in PEAK mode, you will be able to raise the total volume more than in AVG mode, but the waveforms will be more distorted than in AVG mode. In AVG mode it is easier to get a more transparent limiting, but it is also easier to get lots of punch in a drum track since the slower attack time will let the transients through. The original manual recommends the AVG position except when you have very transient sounds.

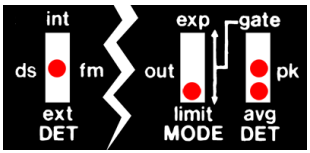
The limiter will let the fastest transients slip through, especially in AVG mode. These transients will be caught by the output clipping, and if they are very short in duration, the effect of the clipping will be inaudible.

Level Limiting (INT, LIMIT, AVG)
Peak Limiting (INT, LIMIT, PEAK)

Ratio = Inf:1, Automatic make-up gain during limiting. **Range** control forced to 60 dB on hardware unit, but active in the plug-in.

AVG **mode is good for transparent limiting**, or added punch to a bass line or drum track.

PEAK **mode is good for hard limiting**, for example adding length to a snare drum sound or to limit fast transients.



De-Essing and Classic Limiting With Pre-Emphasis Filtering

Mode: LIMIT
Detector Type: AVG/PEAK
Detector Source: DS-FM

Threshold: Adjust to material, normally about 6—10 dB of gain reduction.
Release: Usually short, 0.05-0.5 sec
Range: Usually 60 dB
Output: Adjust to material

This is a setting that's very useful for vocal de-essing, but also whenever you need to limit high frequency sounds (such as controlling cymbal levels).

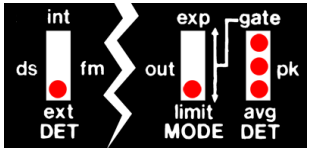
The DS-FM filter only boosts the high frequencies on the signal that enters the detector. The main signal (the one you can listen to) is not affected by this filter.

De-essing (DS-FM, LIMIT, AVG)
Limiting High Frequency Sounds (DS-FM, LIMIT, PEAK)

Ratio = Inf:1. Automatic make-up gain during limiting. High frequency EQ inserted in detector path. **Range** control forced to 60 dB on hardware unit.

AVG **mode: Good for vocal de-essing**. The original manual says: *"It is important to note that the use of Linear Integration Detection [ie. the AVG mode] provides a much more effective mechanism for de-essing than does the conventionally used peak detection."* The reason for this is that the AVG mode's slower response time won't limit transient high frequency sounds (like a 't'), but will limit longer high frequency sounds (like in 'sss').

PEAK **mode: Good for drum track limiting**, if you want to limit the cymbal sibliants. Setup a good limiting without using the DS-FM mode, and then toggle between INT and DS-FM to hear which one you like the best.



Side-Chain Limiting/Ducking

Mode: LIMIT
Detector Type: AVG/PEAK/GATE
Detector Source: EXT

Threshold: Setup the threshold level according to your side chain source level.
Release: Tune by ear. Classic voice-over usually needs longer release times than ducking strings ducking under a 4/4 bass drum.
Range: Use it! Set it to the desired maximum amount of ducking (in dB).
Output: –

A rather normal procedure is to let one track "duck" below another. For example ducking a synth pad by the strike of the bass drum (good use for the GATE mode). Another classic example is using it to duck background music below a voice-over (good use for the AVG mode).

Operating procedure (from the original manual): *"In setting up for the Ducking operation, one would normally start with the Range at maximum (CCW) the Release time at minimum (CCW), and the Threshold at maximum (CW). After applying both signal sources, the Threshold would be adjusted such that a reliable full duck were achieved on each external signal passage. The operator [that's you!] would then adjust the Range control for the desired depth of duck, and the Release control for the desired recovery rate."*

AVG/PEAK MODE: Inverse Envelope Follower (EXT, LIMIT, AVG/PEAK)

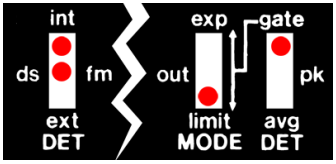
Signal gain is determined by the level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 1 dB decrease in signal gain.

> AVG mode gives the "nicest" and "friendliest" ducking mode. PEAK mode gives a faster ducking that might crackle for some settings.

GATE MODE: Hard Ducking (EXT, LIMIT, GATE)

A 1 dB increase of the side chain signal level (over **Threshold**) causes a 20 dB decrease in signal gain. PEAK detection.

> **Extreme ducking**. Each time the side chain signal goes just a teeny bit above **Threshold**, the main signal will be almost entirely gated. Can be quite effective together with a limited gain reduction, as set by the **Range** knob.



Weird Limiting

Mode: LIMIT
Detector Type: GATE
Detector Source: DS-FM or INT

In the category "Weird Limiting" we can find the negative ratio stuff. The negative limiting mode is a very special case, and is normally used together with longer release times and the **Range** control. In some settings, specifically with long release times and a **Range** at about 10–15 dB, you can get a very crackling sound. Increase the **Range** or the **Release** time to avoid this.

Negative Limiting for "Organ Effects" (INT, LIMIT, GATE) Modified Negative Limiting (DS-FM, LIMIT, GATE)

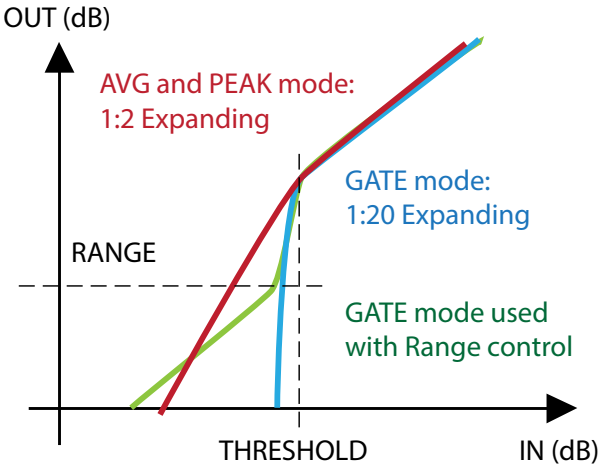
Ratio = 1:-20. As input signal exceeds **Threshold**, a 1 dB increase causes a 20 dB decrease in output level.

> This is the mode that is the hardest to find some use for. It crackles and pops a lot, and one might suspect when reading the original manual that this is a mode that wasn't planned when designing the gear. More like "Wait, what happens if you use the gate together with the limiting mode? Ahh..."

Expanding Modes

The expansion modes is operated in a very similar way to the limiting modes, with the big exception that it is, well, expanding rather than limiting.

Another exception is that the GATE mode actually gates the signal in the EXP mode, while it in the LIMIT mode acts like a negative ratio limiter.



Classic Expansion/Noise Gating

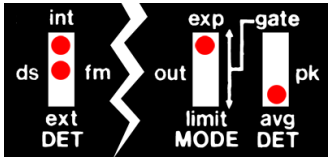
Mode: EXP
Detector Type: AVG/PEAK/GATE
Detector Source: INT/DS-FM

Threshold: Adjust so that the desired sound levels extinguish all lights on the Gain Reduction LED array.
Release: Start with minimum setting, and then adjust "by ear".
Range: Start with 60 dB, adjust to decrease the effect caused by the gating.
Output: Usually below 0 dB

In the classic noise gating mode you have three different settings of the **Detector Type**: AVG, PEAK and GATE.

The GATE mode is the most brutal mode, and is pretty efficient to use on drums or heavy metal guitar if you want a more aggressive sound, but it should be regarded more as a creative tool rather than a problem solver.

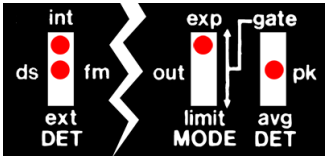
In all of these modes you can toggle between INT and DS-FM to get different weighting of the frequencies. Setting the **Detector Source** in DS-FM mode will make the expander trig more easily on high frequencies, which can be a good thing if you for example are gating a drum beat and want to avoid gating the high frequency cymbals. Since noise often have a lots of high frequency content, you should set it to int if you want to do actual noise gating. But you probably want to use it more creatively anyway...



AVG MODE: Soft Noise Gating (INT/DS-FM, EXP, AVG)

Ratio = 1:2. A signal *x* dB below **Threshold**, will be attenuated another *x* dBs.

> The least obtrusive noise gate. Good for classic noise gating with sources that doesn't have strong transients (such as voice and strings). Use on slow to medium attack sounds.

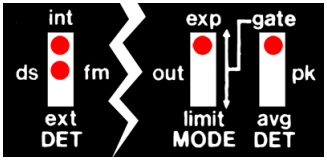


PEAK MODE: Soft Noise Gating with Faster Attack Time (INT/DS-FM, EXP, PEAK)

Ratio = 1:2. A signal *x* dB below **Threshold**, will be attenuated another *x* dBs.

If the attack of the instrument is moderate to fast, such as with drums or certain types of guitar, it is better to use this PEAK mode. The faster attack time in PEAK mode assures that the gain is fully restored when the transient strikes.

> This mode can be pretty nice to use if you want to shorten the decay of a sound, such as a bass line or steel stringed acoustic guitar. With this mode you can get a standard Jazz Bass to sound like a damped Hofner string bass, or a steel stringed $3000 Martin to sound like a $30 banjo... Just set the **Release** time so that you get the right amount of decay.



GATE MODE: Hard Noise Gating (INT/DS-FM, EXP, GATE)

Ratio = 1:20, PEAK detection of input signal. A signal *x* dB below **Threshold**, will be attenuated another *x* dBs.

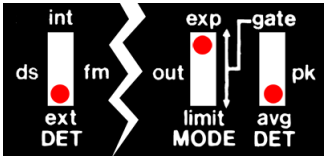
> Use this setting as a traditional, boring hard noise gate, or... use it to create hard hitting beats from almost any drum track! Set a short release time and aggressive gating, and set the **Output** volume high to get some distortion. Stack up many Dyna-mites with different amounts of distortion and release times to get fat snare and bass drum sounds... Toggle the DS-FM switch to include/exclude cymbals.

Keying Modes

Mode: EXP
Detector Type: AVG/PEAK/GATE
Detector Source: EXT

While most people are accustomed to ducking, a.k.a side chain compression—for example ducking a synth pad when the bass drum strikes, the art of keying is still somewhat of a hidden secret for most people. I mean, how many times have you heard Oprah talk about the benefits of keying?

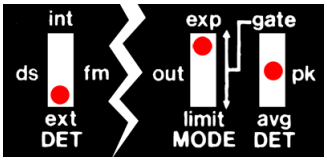
Keying is when you let your main track (say a bass line) follow the envelope of the side chain track (for example a bass drum). So with this example, the bass line will be gated whenever the bass drum is silent, and whenever the bass drum strikes, so will the bass line.



AVG MODE: Envelope Following Soft Keying (EXT, EXP, AVG)

Signal gain is determined by the level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 1 dB increase in signal gain.

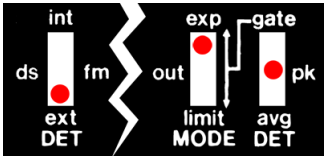
AVG mode has (as usual) a more natural sound, but will slow down the attack of the main signal. Use when you don't need a fast attack time when opening the gate.



PEAK MODE: Envelope Following Soft Keying with Faster Attack (EXT, EXP, PEAK)

Signal gain is determined by level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 1 dB increase in signal gain.

The extremely fast attack time will assure that the keyed signal has full gain when the attack of the side chain signal arrives, but this can sometimes cause the transient to "pop" or "crack" when the gate opens. This is usually not a problem if your mix consists of both the keyed and the side chain signal, since the real transient from the side chain signal will be audible at the same time as the (unwanted) "pop" sound.



GATE MODE: Hard Keying (EXT, EXP, GATE)

Signal gain is determined by level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 20 dB increase in signal gain. PEAK detection.

Same fast attack times as the Soft Keying mode with PEAK detection, but a much more aggressive on/off type of gate. Before the age of the DAW, it was very useful when you needed to tighten up poorly performed background vocals or horn sections. Use the player with the best timing as key, and use the Hard Keying mode on the other players to assure that all off-time notes are being gated. Today, it's just a cool effect.

Mono and Stereo Operation

Inserting the Dyna-mite in STEREO mode makes it behave just as the real unit in "STEREO COUPLE" mode.

Buying Recommendations

The Valley People Dyna-mite is a very cool and affordable hardware limiter/expander. Many people don't know how to use it and say they don't like it just because they cannot operate it. Once you've learned the plug-in you will have no problem at all to operate the real hardware, so if you find a unit—make sure you pick it up!

(The thing on top of the unit is a description of all the different modes, the same texts that are displayed in our nifty little display.)

Credits

Oscar Öberg – modeling, **Niklas Odelholm** – graphics programming. **Torsten Gatu** – framework programming. **Arvid Rosén** – modeling and framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by Valley People Incorporated.

ALL VISUAL AND AURAL REFERENCES TO THE VALLEY PEOPLE DYNA-MITE ARE TRADEMARKS BEING MADE WITH WRITTEN PERMISSION FROM PMI AUDIO. THE VALLEY PEOPLE, DYNA-MITE AND ASSOCIATED LOGOS ARE TRADEMARKS OF PMI AUDIO GROUP, USED UNDER LICENSE. ALL SPECIFICATIONS SUBJECT TO CHANGE WITHOUT NOTICE. ALL RIGHTS RESERVED.

63 Valley People Dyna-mite Slam and Gate

The original Dyna-mite by Valley People is iconic for both good and bad reasons. It was built to be a Swiss Army knife of dynamics, with a gate, expander, peak limiter and RMS limiter all built in to a single unit. It excelled at these tasks, being both the rawest and punchiest compressor out there, as well as the smoothest, most care-free gate/expander there is. That's one reason it's iconic.

The other reason is its terrible user interface. Engineers remember the Dyna-mite as "the thing that sounded awesome but was too scary to use on a session". Trying to cram that much functionality into a unit that was 1U high and half-rack wide made it very difficult to use. The combination of three 3-way switches also led to weird combinations, such as the -20:1 negative limiting setting. What's with that?

But that shouldn't take away the fact that the Valley People Dyna-mite sounds amazing. It can make a snare punchier than anything else, it can totally squash a drum kit, it's really (really!) a really good gate that you should use on your toms, and it can create tons of cool sounds. Too bad no-one dared to use it.

Fast forward to 2019. The futuristic AI-blockchain engineers work frantically with their 3D holograph "Minority Report" inspired Emacs terminals. They see the solution. A new user interface for the holy foundation of the Dyna-mite. It would not be sacrilegious, it won't change the essence of the divine circuits, merely it's face. They can do it, they will do it!

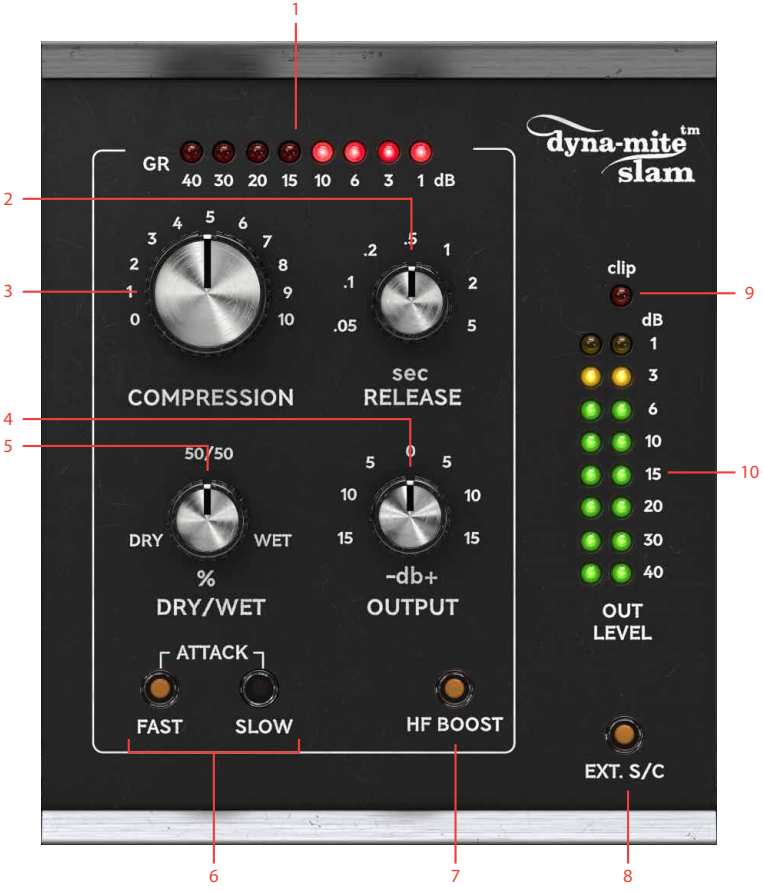
Dyna-mite Slam

Slam is the compressor/limiter part of the original Dyna-mite. The only thing that has been added is a parallel compression knob, because you know you want it. Away with the Range knob, and all the weird settings. This is as close to a one-knob compressor as Dyna-mite will ever become.

For more in-depth information on the Dyna-mite's operation, please see the Dyna-mite chapter.

User interface

There's one knob that's bigger than the others. That's the first one you should reach for.



1. Gain reduction meter
2. Release knob
3. Compression knob
4. Output knob
5. Dry/Wet knob
6. Attack Fast/Slow
7. HF Boost
8. Ext. S/C
9. Clip meter
10. Output meter

Parameters

Compression Increase Compression for more compression. This corresponds to the Threshold knob on the Dyna-mite, but backwards. Turn it up for more, down for less.

Release Sets the release time of the compressor. Shorter release means more aggressive pumping, a slow release makes it smoother.

Dry/Wet Decrease to blend in some of the unaffected dry signal. This is very useful when applying heavy compression, to mix some of the original transients back into the audio.

Output Sets the output volume. The Dyna-mite has automatic make-up gain, so you probably don't have to reach for this knob that much. Don't be afraid of overdriving the output, the built-in distortion is thick and funky, and very useful!

Attack Fast/Slow These two buttons adjust the attack time. They correspond to "pk" and "avg" on the original Dyna-mite.

HF Boost The HF Boost increases the high frequencies that the compressor's detector sees, which makes it trigger more easily for high frequencies. If Dyna-mite Slam reacts too much on low frequency content, such as the kick drum on a drum bus, engage HF Boost to make it listen more to the top end. With HF Boost engaged, the compressor will react more naturally to the frequency content, since its internal detector will "hear" the sound more like our ear would hear it.

This corresponds to the "ds/fm" mode on the original Dyna-mite.

The HF Boost does not affect the audio path we listen to, just the audio that the detector uses to decide if the compressor should reduce gain or not.

Ext. S/C Engage the external sidechain (if available). The external sidechain disables the automatic make-up gain.

Gain Reduction Meter Shows the amount of gain reduction.

Output Meter LED meter showing the output volume.

Clip Meter The Clip Meter will turn on if the Dyna-mite is clipping the output signal. The output clipping has been carefully modeled from the original unit and can be used as a distortion effect.

In use

Slamming It

If you need to get something to happen with your drum bus, you might need to slam it. Set Attack to FAST, Release to .1 and increase Compression until you read 10-20 dB of gain reduction. Decrease Dry/Wet until you get enough of the original drums back into the audio.

Distorting It

The output will distort at roughly 0 dB, and with compression and auto make-up it might be difficult to hit that level. But there's a trick! If you enable "Ext. S/C", you will disable the auto make-up, and you can turn up Volume to get to that sweet distortion!

Punchy Drums

Punchy means a fat, energy-filled, attack with a lot of mass. That's what the Slow button does.

Click on Slow and increase Compression until you get 6-20 dB of gain reduction. You should hear how the first part of the sound (the transient) gets more attack while the rest of the sound gets lower in volume.

Use it on a snare drum to make it sound like the drummer is hitting the drum harder.

Dyna-mite Gate

The Dyna-mite Gate is a two-in-one expander and gate. Its "auto release circuit" makes it very musical, and it's as close to a set-and-forget gate as we've ever seen. The knobs should be pretty standard, but the sound is something more than just standard.

For more in-depth information on the Dyna-mite's operation, please see the Dyna-mite chapter.

User interface

As with Dyna-mite Slam, the Threshold knob is bigger than the other knobs because it's the main control. Decrease Threshold so that the gate opens up on your wanted sounds (for example a drum hit), but closes in the silences between hits. Make the release time short, but still slightly longer than the sustain of your sound. If you need total silence between hits, click the "FAST 1:20" button. If you need a smooth gate, click the "SLOW 1:2".

1. Gain reduction meter
2. Release knob
3. Threshold knob
4. Output knob
5. Range knob
6. HF Boost
7. Ext. S/C
8. Clip meter
9. Output meter

Parameters

Threshold Adjust the Threshold to change the sensitivity of the gate.

Release Sets the release time (the time it takes for the gate to close).

Range Sets the maxing amount of gain reduction. If you only need 10 dB of attenuation, set Range to 10 dB.

Output Sets the output volume.

Fast Gate Hard gating (1:20) with a fast attack (time it takes for the gate to open). Use for percussive sound where maximum attenuation is needed, with the risk of making the gate sound slightly artificial. This corresponds to the "gate" mode on the Dyna-mite.

Fast Exp. 1:2 expansion with a fast attack (time it takes for the gate to open). Use for most percussive sounds, such as drums, to achieve a natural sounding gate. This corresponds to the "pk" mode on the Dyna-mite.

Slow Exp. 1:2 expansion with a slow attack (time it takes for the gate to open). Use for sounds with a soft onset, such as strings, pads, etc. This corresponds to the "avg" mode on the Dyna-mite.

HF Boost The HF Boost increases the high frequencies that the gate's internal detector sees, which makes it react easier to high frequency sounds. If you need to gate a kick drum, it's a good bet to leave it off. If you need to gate a snare with high frequency content you can turn it on to achieve better separation between kick and snare.

This corresponds to the "ds/fm" mode on the original Dyna-mite.

The HF Boost does not affect the audio path we listen to, just the audio that the detector uses to decide if the gate should open or not.

Ext. S/C Engage the external sidechain (if available).

Gain Reduction Meter Shows the amount of gain reduction.

Output Meter LED meter showing the output volume.

Clip Meter The Clip Meter will turn on if the Dyna-mite is clipping the output signal. The output clipping has been carefully modeled from the original unit and can be used as a distortion effect.

In use

Standard toms gating

Set the switches to "Fast Exp.". Adjust Threshold to taste. (Short tip, but it really is that easy.)

Dyna-mite Overdrive

Set Threshold to minimum. Increase Output Volume until the clip LED turns on. Compensate for the loud output signal by turning down the fader in your DAW.

Sub-Synth Sidechaining

This is a neat trick from the past, that use the sidechain to key a synth so that the synth sounds whenever the sidechain has signal. You probably have plug-ins that do all this, but it is cool to learn how it's done.

Setup the Dyna-mite Gate on a synth track that plays an extended low note. Send the kick drum to the sidechain of the Dyna-mite Gate, so that the gate opens whenever the kick drum hits.

The synth will only sound when the kick drum sounds, and you can use the synth to play a low note that harmonize with the kick drum. Adjust the release time to set the sustain of the note.

Tighten Back-up Harmonies

Okay, another side-chaining trick, but keying is the new black. Setup the Dyna-mite Gate on the backing vocal bus. Sidechain it with the lead vocals. Let the switches be "Slow Exp." with a fairly fast Release time. Adjust Threshold so that the gate opens whenever the lead vocal sings.

Credits

Niklas Odelholm - DSP and UI programming, graphic design, user manual. **Oscar Öberg, Arvid Rosén** - modeling**. Ulf Ekelöf** - 3D rendering.

64 Vintage Amp Room

Introduction

STRAIGHTFORWARD AND SIMPLE, Vintage Amp Room emulates three great guitar amps in a complete studio setup with speaker cabinets and fully flexible microphone positioning. It's easy to use for live performances and recordings, as well as for reamping and lastminute tweaking.

Amp Room has none of the added gadgets or weird-sounding presets sometimes found in simulation software. Focusing on quality and accuracy, it simulates three amps, with sounds that are timeless and authentically raw. The goal is not to give you a preproduced, ultimate guitar sound, but to provide you with the ultimate tool to create your own. Because after all, only you know exactly what sound you want.

Product Philosophy

EVERYTHING ABOUT THIS PRODUCT is authentic. The raw and dynamic sound is an absolute replica of what you would get if you had these amps set up in a real studio. What you see on the screen are photorealistic 3D renderings of the complete setups, and you get to move an actual mic around when deciding which position is best, from near field offaxis to far field and back, continuously, without preset positions.

THE IDEA IS THAT using Amp Room should be like working directly with these amplifiers set up in a real studio, with a microphone in front of the cabinet, connected to your DAW. And we kept it simple. Amp Room has no knobs with dubious or unintuitive functionality, and no added gadgets or ridiculously fake-sounding effects.

The sound of the amps has not been tampered with, because we don't give you a pre produced, ultimate guitar sound. We simply provide you with the tools: three great, classic amplifiers, in digital form. Then it's up to you to add compressors, EQs, reverb, or any other effects, to get the sound you want.

In short, you need the same skills to master this software as you need when dealing with a real studio setup. You can get back to doing what you do best, because music production is about using your ears, not your computer skills.

Supernormalize

The "supernormalize" feature (from the beginning the internal name of a slightly magic component of our model building toolbox), makes sure that the output signal from Amp Room always has a reasonable volume. From a user perspective, this means that the Amp Room never clips the signal (unwanted digital distortion). Also, the output volume is normalized, so even with low settings on the Volume knob in Amp Room, the output volume will be within a reasonable level and ready for digital recordings. Compare this to the real amp, which easily can have a dynamic range of 100 dB.

From a technical perspective, this means that the signal path within Amp Room always has the highest dynamic range possible, and you won't lose any bits on the way.

User Interface

Simplicity has been the goal when we developed this product; the amps and mics should work just as they do in the real studio. Unfortunately, moving a mouse cursor on a 17" LCD screen isn't exactly like grabbing the real mic stand in a cozy studio. Nevertheless, we have tried to imitate that feeling as far as it is possible, and if you have experience of working in real studios, you will notice that Amp Room handles and sounds the way you expect it to do.

Amp Panel (Top Area)

In the Amp Panel you can alter the settings of the amp by clicking the knob and dragging the mouse up and down. Switches will be switched either by clicking on them, or by click-and-dragging the mouse up and down.

Amp Panel

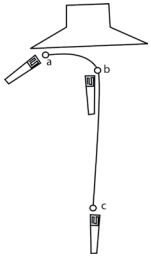
Room View

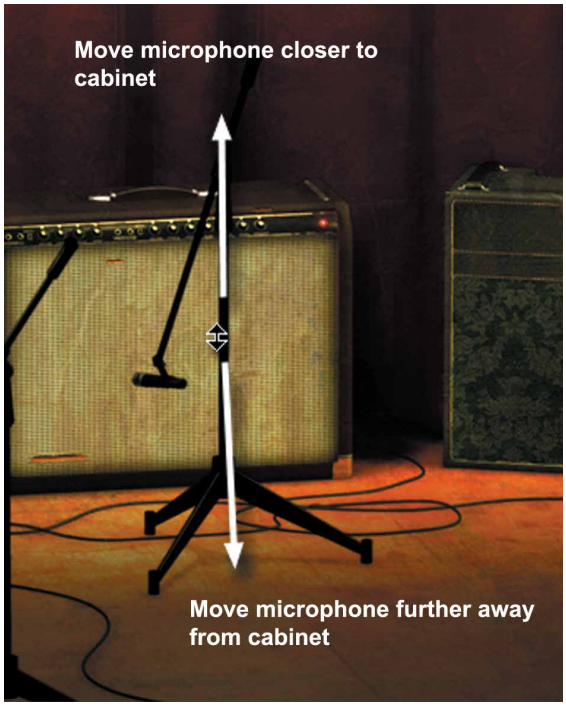
Room View (Bottom Area)

In the Room View you can do two things, select one of the other amps or change the mic position.

Mic Position

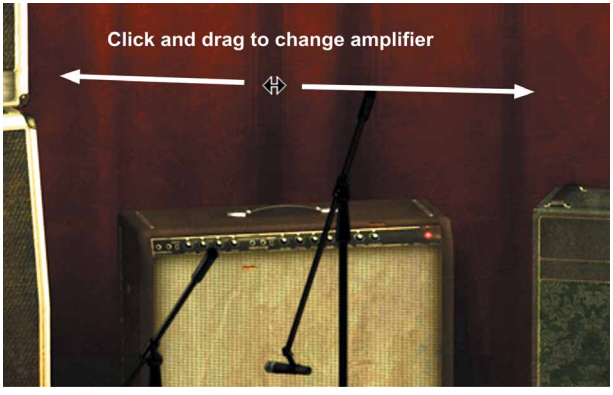
When the mouse is located over the mic stand, the mouse pointer changes to an "up-down" arrow. Click and drag the mouse up or down to change the position of the mic stand. The microphone moves along a predetermined path, so you only need to move the mouse up or down, just as if you were changing a knob.





Amp Selection

You can change amp by clicking on the background and dragging the mouse to the left or right. The mouse pointer becomes a "left-right" arrow when you are pointing at the back ground to indicate that it is possible to change amplifier.



White Amp

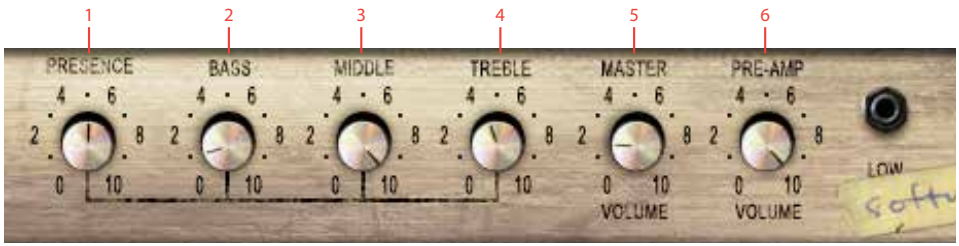
White is based on an allround great amp, a classic that set the industry standard for classic, distorted rock'n'roll sounds. The genius is its simplicity. It's stripped down to the bare essentials – all tubes, only one channel, and no reverb or any other built-in effects.

Soundwise, it goes from mad screaming preamp shred, to warm, speakersabouttobreak power amp distortion. Perfect when you need a characteristically distorted edge and a powerful roar.

First Use

Set all parameters in the middle (12 o'clock). Turn down the **Master Volume** a little bit (to 10 o'clock) and play your guitar while changing the **Pre Amp Volume**. The sound goes from almost HiFi clean on low **Pre Amp Volume** settings, to a crunchy distortion (**Pre Amp Volume** in the middle), and finally to a high-gain setting with **Pre Amp Volume** on max. If you want an even meaner distortion, turn down the **Middle** and turn up the **Master Volume**.

The sound will change quite dramatically when you start moving the mic away from the cabinet, since certain frequencies will cancel each other out due to interference between the four speaker drivers.



1. Presence
2. Bass
3. Middle
4. Treble
5. Master volume
6. Pre-amp volume

Knobs

The functionality of the knobs on the front panel of the plugin correspond exactly to the front panel of the real amplifier.

Presence The presence is used to control the feedback loop and thus the amount of treble in the power amp. Turn it up to get a high frequency boost.

Bass, Middle, and Treble: These knobs are the tone control of the amplifier. This circuit is located after the preamp distortion and is used to shape the sound of the preamp distortion.

Master Volume: This knob controls the amount of power amplifier distortion. The power amp distortion is usually a little bit dirtier than the preamp distortion and gives a crunchier sound that sounds amazing for chords. Set the **Pre Amp Volume** to a low setting and turn up the **Master Volume** to the max to get that typical power amp distortion.

Pre Amp Volume: This knob controls the amount of preamplifier distortion, but since a high output from the preamplifier also makes the power stage distort, this also affects the power amplifier distortion. The preamp distortion is typically much "tighter" than the crunchy power amp distortion. To get a tight preamp distortion, you'll have to turn down the **Master Volume** while maxing the **Pre Amp Volume**.

Brown Amp

Brown is based on one of the most versatile classic amplifiers in music history. It's been used in all kinds of genres, and with all kind of instruments, thanks to a sound that is clean, crisp and clear, but also warm, thick and punchy.

Brown can produce anything from the clearest countrystyle sound for steelguitars to a dirty, bluesy guitar sound. It works for both funky rhythms and mellow electric piano chords. It's got a notoriously warm sound with a massive bass and a shimmering treble. Versatility embodied, it can do more or less whatever you want it to do. Except high gain heavy metal. And baking pizzas.

First Use

Set all knobs in the middle (at 12 o'clock). Set the microphone in a far field position (click the mic and drag the mouse downwards). Turn on the **Vibrato** and **Bright** switches. You should hear a bluesy sound with lots of room ambiance and a nice transition between distortion and a clean sound. Change the **Volume** to get more or less distortion and move the microphone closer to the cabinet to get a "tighter" sound with more bass frequencies.

1. Bright
2. Volume
3. Treble
4. Middle
5. Bass
6. Vibrato On/Off
7. Vibrato speed
8. Vibrato intensity

Knobs

The functionality of the knobs on the front panel of the plugin corresponds to the front panel of the real amplifier, except the spring reverb, which hasn't been included in this plug in. We have also added a "Vibrato" switch, which on the real amplifier is a foot switch. There is also a difference when all the tone controls are set at 0, see "Treble, Middle, and Bass" below.

Volume This amplifier doesn't have a Master Volume control, so the **Volume** knob controls both the preamplifier and power amplifier distortion.

Bright Turn on this switch to get a high frequency boost. Works only at low to moderate **Volume** settings.

Treble, Middle, and Bass These knobs are the tone control of the amplifier, but they also control the amount of distortion. If you, for example, have too much distortion in the bass frequencies, try turning down the **Bass** knob. (Electrically speaking, the tone controls are located before the preamp distortion.)

The tone controls behave exactly like the real thing, except when all the controls are set fully counter clockwise. On the real amp, the amplifier would have become silent, as if you turned down the volume. A lot of reasons made us exclude this "feature," so now when you turn the **Treble**, **Middle** and **Bass** knobs fully counter clockwise, that actually corresponds to the knobs being set to a little bit over nothing on the real amp.

Vibrato, Speed and Intensity Turn on the vibrato (which is in fact a tremolo) by turning on the **Vibrato** switch. Use **Speed** to control the speed of the tremolo, and **Intensity** to control the amount of the tremolo effect.

Green Amp

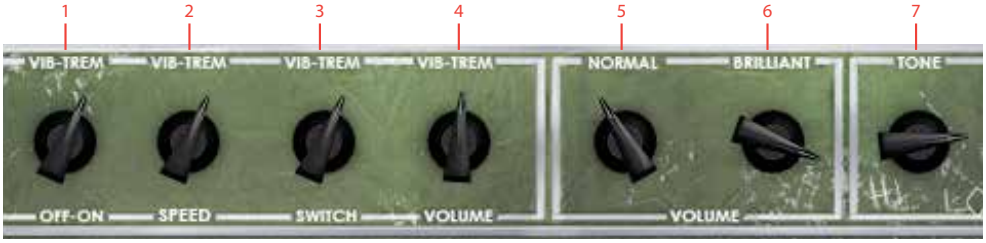
Green is packed with character and produces a pleasant guitar sound with warm yet intense poweramp distortion. Its edginess makes it shine through in mixes, without drowning out other instruments.

Green is based on a famous britpop amplifier from the middle of the 60's. It has three different channels ("Vib Trem", "Normal" and "Brilliant") with two inputs each, and normally a guitar player will connect the guitar to the "VibTrem" input and patch the signal to the other two channels. We have simulated the amplifier exactly this way; the guitar connected to the "VibTrem" input and patched to the other two channels.

To complicate things further, this amp doesn't have a normal tone stack (EQ) but a single tone control which attenuates high frequencies when turned clockwise (different to what one would expect of a tone control). The previous owner of the original amp was so kind as to carve out "HI" and "LO" in the metal panel so that he would remember this. We have of course included this feature in Amp Room.

First Use

Turn on the vibrato/tremolo effect (**Vib-Trem Off-On**) and select the tremolo effect (**Vib-Trem Switch** pointing at "TREM"). Set the **Vib-Trem Volume** knob in the middle position and all other knobs in the minimum position. Now adjust the **Brilliant Volume** to get a good balance between the tremolo effect and a clean sound.



1. Vib/Trem Off/On
2. Vib/Trem speed
3. Vib/Trem switch
4. Vib/Trem Ch. volume
5. Normal Ch. volume
6. Brilliant Ch. volume
7. Tone

Knobs

The functionality of the knobs on the front panel of the plugin corresponds exactly to the front panel of the real amplifier, except that we have added the foot switch as a "Vib-Trem Off-On" switch.

Vib-Trem Off-On Turn on the vibrato or tremolo effect by moving this switch clockwise.

Vib-Trem Speed Three way switch that sets the speed of the vibrato or tremolo effect.

Vib-Trem Switch Select between a vibrato and a tremolo effect. Unlike the makers of the Brown amp, the guys who built this amp actually knew the difference between a tremolo effect (amplitude modulation of a signal) and a vibrato effect (frequency modulation of a signal). Although this vibrato effect doesn't sound anything like a real-world vibrato (opera singerstyle vibrato), it's a lovely effect that gives the sound that special touch.

Vib-Trem Volume Volume control of the Vib-Trem channel. This channel is basically the only one of the three channels that has any preamp distortion worth talking of.

Normal Volume Volume control of the Normal channel. Except for the fuzz-like poweramp distortion on higher volume settings, this channel is very clean and doesn't change the guitar sound so much.

Brilliant Volume Volume control of the Brilliant channel, which has a much brighter sound than the Normal channel.

Tone The Tone knob cuts high frequencies when it's turned clock wise.

Use the **Vib-Trem Volume** to get the distortion you want, and then use the **Normal** and **Brilliant Volume** as a bass and treble control. If you want a more fuzz-like distortion, use only the Normal and Brillant channel.

Bypassing Amps or Cabs

You can choose to bypass the amp or the cabinet by selecting AMP BYPASS or CAB BYPASS from the small box in the lower right corner.

This is very useful if you want to use Vintage Amp Room with an external speaker (or speaker plugin) or if you just want to add a cabinet simulation to a track, such as a recorded line out from your amplifier.

> By bypassing the cabinets in Vintage Amp Room, you can use the cabinets from the other Amp Room plugins. Just insert, for example, Bass Amp Room (with bypassed amplifier) on the track after Vintage Amp Room (with a bypassed cabinet).

Buying Recommendations

If you like the sound from Amp Room and would like to get that sound using the real deal (let's face it, a real vintage amp is always sexier than a plugin), here are some buying recommendations.

White

If you like the sound of the White amp, we recommend you to buy a Marshall JCM800 2203 with a 1960A 4x12" cabinet. Nothing beats the roar from a 100W valve amp.

Brown

Likewise, if Brown is your favorite, we strongly recommend a 1966 Fender Twin Reverb combo with a 2x12" cabinet.

Green

Nothing beats the real stuff, and that quirky vibrato calls for a Vox AC30/6 Treble from the middle of the 60's, perhaps suited with a pair of new Celestion Blue drivers.

Room

And finally, if you like the sound of the room in which the cabinets we're measured, you better call Soulmine Studios in Linköping and book some studio hours. Just tell them to set things up just as they did for the guys from Softube, and you'll be fine from there...

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Credits

Oscar Öberg – concept, preamp modeling, sound and DSP-programming. **Niklas Odelholm** – cabinet and microphone modeling and sound design. **Arvid Rosén** – power amp modeling and sound programming. **Torsten Gatu** – interface and DSP programming. **Per Connman** – amp selection and modeling. **Anders** – graphics programming. **Ulf Ekelöf** – graphics programming and 3D rendering. **Micko** – 3D rendering. **Papa Bear** – collages and graphic profiling.

65 Weiss Compressor/Limiter

Weiss Compressor/Limiter is a versatile mixing and mastering unit based on the Weiss DS1-MK3 and has advanced features like FFT display, waveform display auto release and parallel mix. The Compressor/Limiter is was created to take the gold-standard DS1-MK3 experience to a new level of usability and workflows.

Differences between Weiss DS1-MK3 and Compressor/Limiter.

	Comp/Limiter	DS1-MK3
Scalable FFT display	Yes	No
Hi-Res waveform display	Yes	No
Filters available	• Full band • Band Pass • Low Pass	• Full Band • Band Pass • High Pass • Low Pass
Operational frequency range (Full band)	20 Hz – 20 kHz	20 Hz – 20 kHz
Operational frequency range (Band Pass and Low pass)	20 Hz - 3520 kHz	20 Hz - 20 kHz
Auto release offset	Yes	No
Parallel processing	Dry/Wet balance	Original

Main display features

The main display has several modes: Waveform, Frequency and Knee.

Waveform display

Waveform display is handy for controlling how much of the gain reduction occurs and when in time. It shows the peaks, RMS and gain reduction graphs.

Frequency display

Frequency display is useful to zoom in on a certain frequency when in a band selection mode.

On the Frequency display you can change Frequency, Threshold and Width of the filters. Each parameter has its own click area on the widget. To change the parameter setting, click and drag the selected click area.

Threshold: sets the level which the envelope signal has to reach before gain reduction is applied.

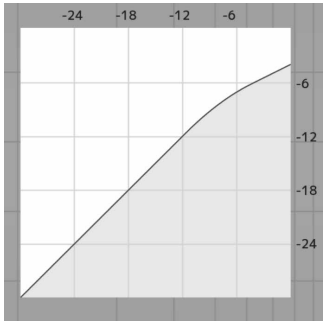
Frequency: sets the center frequency of the filter.

Width: sets the width of the filter.

Knee Display

Knee display shows the knee shape of the compressor.

Soft-knee: Determines how much the knee is rounded. The maximum setting (1.0) chooses a curve that reaches from 0dBFS down to twice the threshold value. This implies that if soft-knee is engaged the signal is already affected even if its envelope lies below the threshold.



Meters

Input Gain Shows Peak and RMS levels.

Output Gain Shows the peak and RMS levels

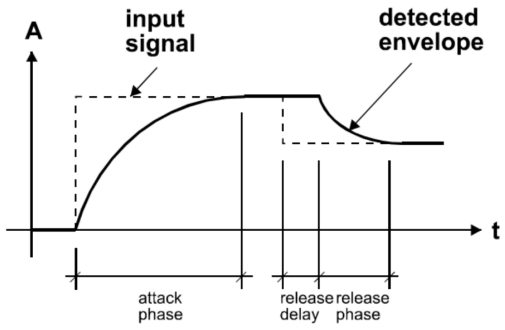
Gain Reduction Meters In stereo mode, the gain reduction meters show gain reduction for L and R channels per band while in Mid/Side mode, the gain reduction meters show gain reduction for M and S.

Parameters

Ratio Sets the slope of the segment above the threshold. The ratio can be adjusted for compression or expansion, depending on the requirement.

Attack and Preview The longer the attack time, the more will the leading edge of fast transients pass by the gain reduction circuit unaltered (as in Graphic 9 during attack phase). Very fast attack time settings such as 20µs, i.e. one sample period, do catch every transient, but may distort low frequencies. To utilise longer attack times and still catch fast transients, use preview.

Release and Average The input signal is monitored with two different methods: peak amplitude and RMS value. The peak amplitude is the programme portion we don't perceive as very loud,



Graphic 1: Envelope parameters

however which can easily cause overloads. The RMS value of the programme material we perceive as loudness. Its variations contribute most to the dynamic range of the audio. The sidechain processor compares the two measurements it obtained from the peaks and the RMS. The ratio between the two determines which time constant would be the appropriate release, e.g. after short duration peak the faster release will be applied during the release phase. The time period over which the RMS value of the programme material is averaged is set by the "average" parameter. The effect of the average parameter can best be studied on the gain

reduction meter. Fast "average" settings will cause most of the gain reduction meter to move very fast (depending on the "release fast" setting) , slower settings will just have the top part of the gain reduction move fast, with the bottom part depending on the "release slow" setting.

Auto release offset allows to control all three release parameters (Release Fast, Release Slow and Average) with one Release knob on the main screen. The feature adds the defined offset between Release Fast, Release Slow and Average. If Auto Release Offset is set to 100 ms and Release Fast is set to 50 ms, Release Slow will be automatically calculated as 150 ms and will change proportionally to Release Fast. TIP: It is possible to turn off the Auto Release Offset feature in the Side Panel. In this case all three release parameters (Release Fast, Release Slow and Average) wil have to be set manually.

Release Delay Normally the release phase begins immediately after the programme material has fallen in level. The release delay determines how long the Compressor/Limiter holds the current level before entering release (see Graphic 9). The resulting dynamic characteristic is very smooth and avoids otherwise typical 'pumping' effects. The ideal setting is programme dependent and experimentation may be necessary to achieve the best combination of all dynamic parameters.

Parallel mix the parallel processing feature is implemented as a DRY/WET mix. Parallel Comp. button activates and disables the parallel processing for instant A/Bing.

TIP: Any move of the Parallel Mix button will enable the parallel processing if it was turned off.

Band Selective Mode In this mode, a frequency crossover is put into the signal path, splitting it up into two or three bands, depending on the filter type setting. One of the bands is compressed, the other is delayed by the same amount used as „delay" in the compressor stage. This mode is completely transparent if no gain reduction is applied.

Filter Type Selects the available filter types: Low Pass and Band Pass.

Ganged /Unganged The Compressor/Limiter allows for independent parameter settings in channels 1 and 2. If the ganged key is lit, both channels operate with the same parameter set. If the ganged key is not lit, the two channels can operate with different parameters. The CH1 and CH2, Mid/Side key assigns the knobs and the display to the appropriate channel.

Mid/Side Mode and Sidechain Link The Compressor/Limiter can be switched to Mid / Side mode. In M/S mode it usually is preferable to set the channels unganged and the sidechain unlinked, Channel 1 will operate as M part and Channel 2 will operate as S part.

Side Panel

On the side panel you can find additional parameters.

Output There are three types of limiters available in Compressor/Limiter plug-in:

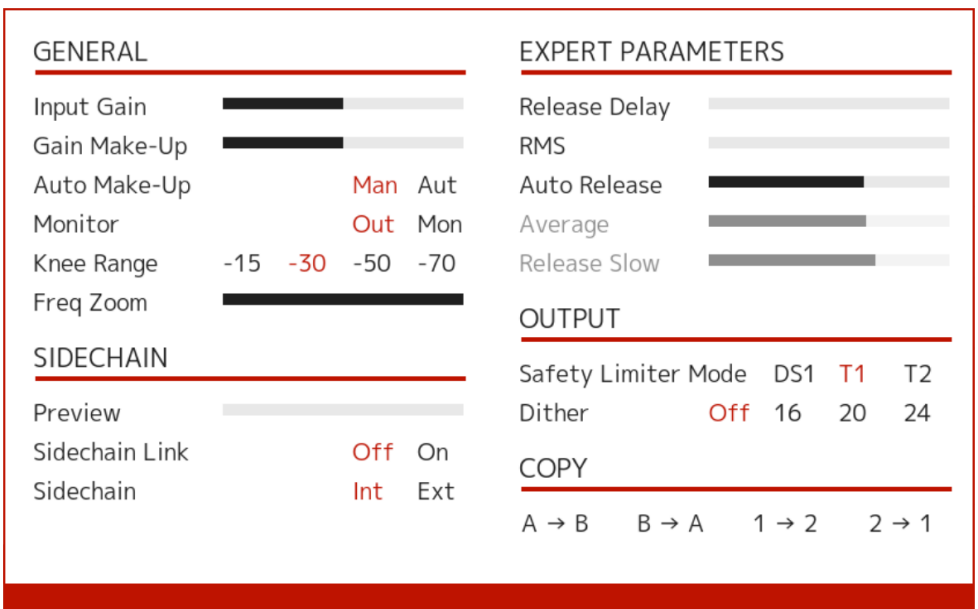
-Original Safe limiter from DS1-MK3 hardware

- New generation Weiss brick-wall Limiter Type 1

- New generation Weiss brick-wall Limiter Type 2 (True Peak)

Limiters Type 1 and Type 2 are designed to provide the most transparent sound for mastering applications performing at high RMS levels.

TIP: Use Type 1 for highest RMS values and Type 2 when True Peak limiting is needed



Monitor In band selective mode turning on the monitor feature allows you to listen directly to the processed band, ideal for zooming in on offending material and listening to the effect of the compressor stage.

Auto Make-up Allows to compensate gain automatically.

Knee range Sets different ranges for the knee view. Set larger ranges when using low threshold settings.

Frequency Zoom Allows to zoom in the FFT display view.

TIP: it is possible to click and drag the grid on the Frequency display to zoom in and out as well.

Sidechain parameter Set the external or internal sidechains.

Dither Sets dithering to 16, 20 and 24 bits.

RMS In the Compressor/Limiter the peak measurement is supplemented with an RMS measurement with variable averaging time. The left endstop of this parameter switches the detection to Peak mode (default one).

Copy option Allows to copy channels and settings from A to B, B to A, and channel settings from Channel 1 to Channel 2 and vice versa in order to have the same parameters in the two channels to start with.

TIP: you may copy or switch the settings from the bottom panel of plug-in as well

Credits

Paul Shyrinskykh - features specifications and product management. **Arvid Rosén** – DSP and framework programming. **Björn Rödseth** – framework programming. **Niklas Odelholm** – graphic design.**Ulf Ekelöf** – 3D rendering. **Daniel Delviken** - marketing. **Maxus Widarson** - quality assurance.

66 Weiss DEESS

Weiss DEESS is a two-band de-esser based on the same algorithms used in Weiss DS1-MK3. DEESS provides features like scalable FFT display, two independent bands for ultimate control, near zero latency mode for real time recordings, additional notch filter design (originally not available in Weiss DS1-MK3), and flexible monitoring options.

Differences between Weiss DS1-MK3 and DEESS

	DEESS	DS1-MK3
Number of operational bands	2	1
Scalable FFT display	Yes	No
Low latency mode	Yes	No
Filters available	• Notch • Band Pass • High Pass	• Full Band • Band Pass • High Pass • Low Pass
Operational frequency range	2 kHz – 20 kHz	20 Hz – 20 kHz
Dithering	No	Yes
Auto Makeup gain	No	Yes
RMS detection	Peak	Peak – RMS
Preview (lookahead)	20 microseconds	Variable
Safe Limiter	No	Yes

Overview

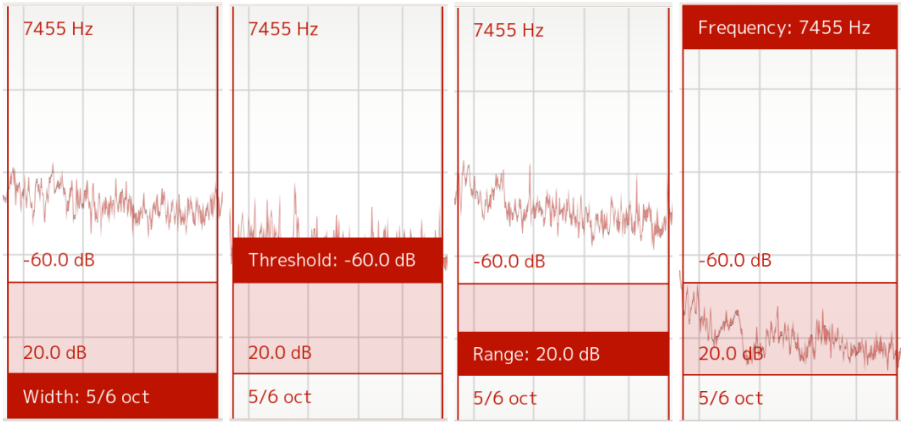
1. Amount of gain reduction
2. Gain reduction meters
3. Output gain
4. Gain reduction values
5. Input gain
6. Side panel

There are two operational bands in DEESS: the Low Band and the High Band. In stereo and Mid/Side mode, the two bands allow to control the mid-range resonances and high-frequency sibilance simultaneously, providing the ultimate control.

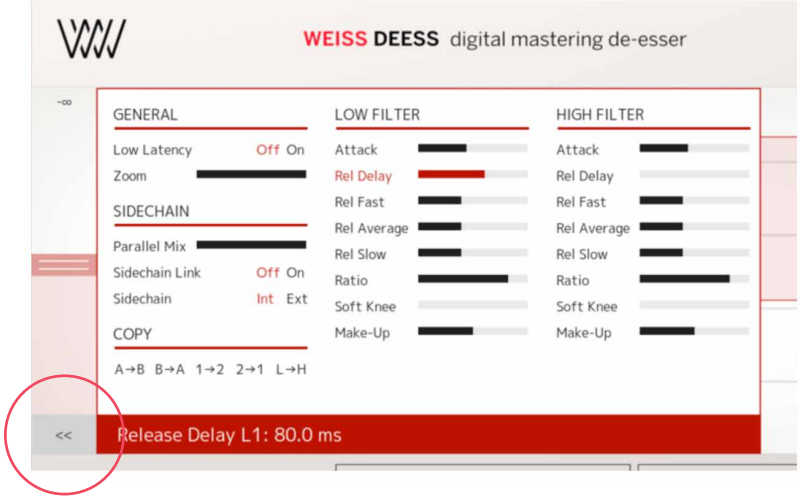
Main display

On the main display you can change Frequency, Threshold, Range and Width of the filters. Each parameter has its own click area on the widget.

To change the parameter setting, click and drag the selected click area.



Options panel is available by clicking the arrows icon at the bottom left corner of the main screen.



Meters

Input Gain Shows Peak and RMS levels.

Output Gain Shows Peak and RMS levels.

Gain Reduction Meters Show gain reduction for Low and High bands. In stereo mode, the gain reduction meters show gain reduction for L and R channels per band while in Mid/Side mode, the gain reduction meters show gain reduction for M and S per band.

Tip: You can monitor the amount of gain reduction and affected frequency range on the FFT display.

Parameters

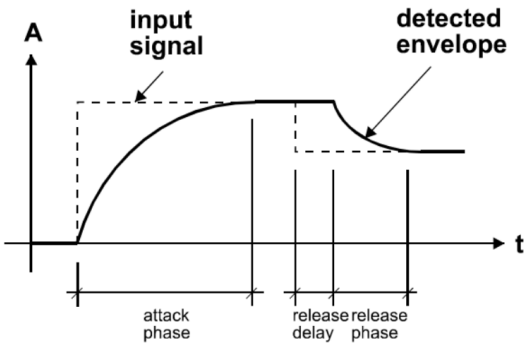
Threshold Sets the level which the envelope signal has to reach before gain reduction is applied.

Frequency Sets the center frequency of the filter.

Range Sets the operational gain reduction range.

Width Sets the width of the filter.

Attack The longer the attack time, the more will the leading edge of fast transients pass by the gain reduction circuit unaltered (as in Graphic 1 during attack phase). Very fast attack time settings such as 20µs, i.e. one sample period, do catch every transient, but may distort low frequencies.



Graphic 1: Envelope parameters

Release and Average The input signal is monitored with two different methods: peak amplitude and RMS value. The peak amplitude is the programme portion we don't perceive as very loud, however which can easily cause overloads. The RMS value of the programme material we perceive as loudness. Its variations contribute most to the dynamic range of the audio. The sidechain processor compares the two measurements it obtained from the peaks and the

RMS. The ratio between the two determines which time constant would be the appropriate release, e.g. after short duration peak the faster release will be applied during the release phase. The time period over which the RMS value of the programme material is averaged is set by the "average" parameter. The effect of the average parameter can best be studied on the gain reduction meter. Fast "average" settings will cause most of the gain reduction meter to move very fast (depending on the "release fast" setting) , slower settings will just have the top part of the gain reduction move fast, with the bottom part depending on the "release slow" setting.

Release Delay Normally the release phase begins immediately after the programme material has fallen in level. The release delay determines how long the DEESS holds the current level before entering release (see Graphic 9). The resulting dynamic characteristic is very smooth and avoids otherwise typical 'pumping' effects. The ideal setting is programme dependent and experimentation may be necessary to achieve the best combination of all dynamic parameters.

Ratio Sets the slope of the segment above the threshold. The ratio can be adjusted for compression or expansion, depending on the requirement.

Soft-knee Determines how much the knee is rounded. The maximum setting (1.0) chooses a curve that reaches from 0dBFS down to twice the threshold value. This implies that if soft-knee is engaged the signal is already affected even if its envelope lies below the threshold.

Gain Makeup Moves the whole transfer curve up or down. In de-essing mode the gain makeup is usually set between the minimum (- 10dB gain) and "off" (0dB gain) to actually attenuate the processed signal.

Low Latency Mode Sets the plug-in in near zero latency mode. The latency @96khz is 51 sample (~0.5 millisecond). Please note: in the low latency mode the Band Pass filter is not available. Low latency mode uses non-linear phase filters. The mode is useful during real time recordings, but for ultimate sound quality and transparency use the original linear phase mode.

Zoom Allows you to zoom in and out to a specific frequency range in the FFT display. TIP: it's possible to click and drag the FFT display area in order to change the zoom

Copy Allows to copy settings from A to B, Channel 1 to Channel 2 and vice versa, and also settings from Low Band to the High Band.

Parallel Mix Controls the amount of Dry (unprocessed) and Wet (processed) signal.

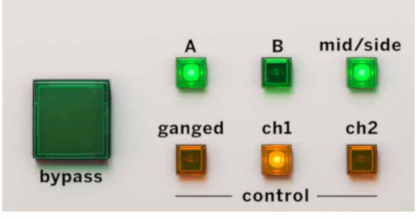
Sidechain Link Links and unlinks the sidechains. It is useful to unlink the sidechains when in Mid/Side mode.

Sidechain Sets the internal and external sidechain

Filter Type Selects the available filter types: Band Pass, Notch and High Pass.

Please note: High Pass filter is available in the High Band only.

Solo and Monitor Provide the monitoring options for the selected band. The Solo allows you to listen directly to the affected frequency band while the Monitor allows to monitor how much of the actual de-essing happens. Please note: Solo and Monitor functions monitor both LR or MS channels simultaneously for the selected band.



On Turns on and off the processing of the selected band.

Mid/Side Enables the Mid/Side processing.

A and B Allow instant switching between the different settings.

Ganged /Unganged The DEESS allows for independent parameter settings in channels 1 and 2. If the ganged key is lit, both channels operate with the same parameter set selected by CH1 or CH2 button. If the ganged key is not lit, the two channels can operate with different parameters.

Bypass Allows you to bypass the plug-in.

Input Gain	Controls the input gain to the plug-in.
Output Gain	Controls the output gain from the plug-in.

In Use

The best way to start using DEESS is to look through the presets first and tweak the thresholds to match the levels of the program material. The presets provide the starting point for many of the de-essing scenarios. For real time recording, use "Low latency tracking" preset as a starting point. In case you need to change the center frequencies of the filters, use Solo to find the offending frequency. Using Zoom to zoom in and out the FFT display can help to find the peaks of the offending frequency as well.

Often you'll need to use two bands for efficient de-essing in order to tame different resonances in vocals. "s" sound resonances usually reside in the 7 khz area, while "ch" and "sh" resonances reside in the area of 4 khz and below.

Tip: you can enable filter band highlight in the setup menu of the plug-in. In some situations, it's useful to see parameters of what band are being edited.

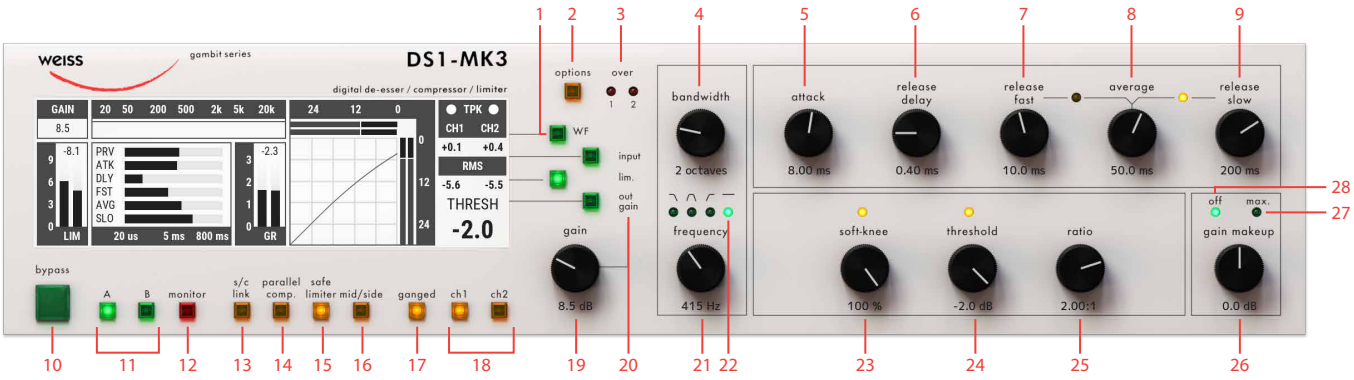
Credits

Paul Shyrinskykh – features specifications and product management. **Arvid Rosén** – DSP and framework programming. **Björn Rödseth** – framework programming. **Niklas Odelholm** – graphic design. **Ulf Ekelöf** – 3D rendering. **Daniel Delviken** – marketing. **Maxus Widarson** – quality assurance. **Igor Miná** – user manual layout.

67 Weiss DS1-MK3

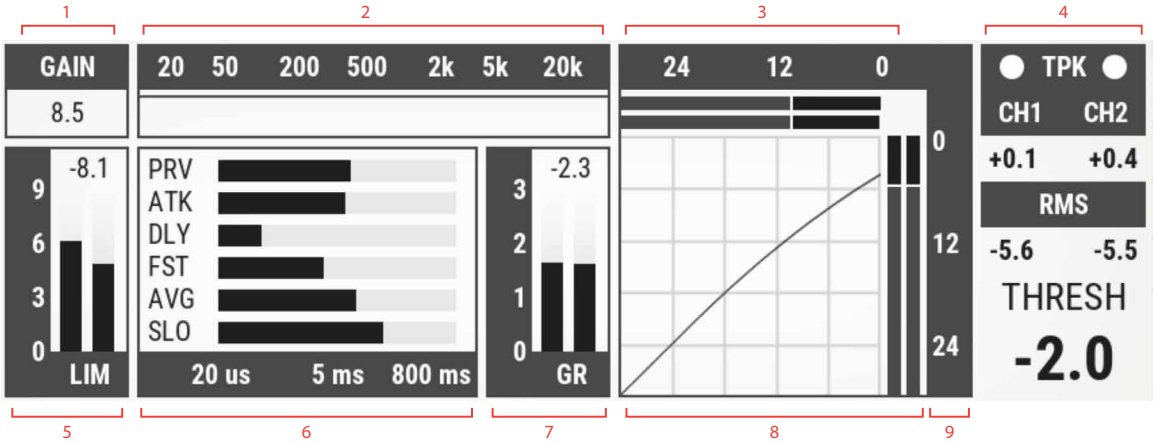
This two channel digital De-esser / Compressor/ Limiter features linear-phase crossover filters covering the whole audio band. The controls for the two channels can be ganged or unganged (useful for M/S compression). Similarly, sidechain linking can be turned off. Double-precision up- and down-sampling units within the DS1-MK3 result in high transparency for absolutely clean sound. The non-linear segment can be adjusted from 1000:1 to 1:5, allowing every kind of dynamic processing, from limiting to upward expansion (for over-compressed signals). Put back a little dynamics into that ultra-finalized mix! An additional safety-limiter Type 1 and Type 2 will allow the most transparent sound for mastering and mixing applications.

User Interface



1. Waveform view
2. Options
3. Overload meters
4. Bandwidth
5. Attack
6. Release delay
7. Release - fast
8. Average
9. Release - slow
10. Bypass
11. A/B settings comparison
12. Processed frequency band monitor (in band split mode)
13. Side-chain link
14. Parallel compression
15. Safe limiter
16. Mid/Side
17. Ganged
18. Channel 1/Mid and Channel 2/Side
19. Gain
20. Input gain, Limiter gain and Output gain
21. Center frequency
22. Filter type
23. Soft-knee
24. Threshold
25. Ratio
26. Gain makeup
27. Auto gain compensation
28. Gain makeup off

Screen



1. Currently selected gain
2. Crossover settings
   (Frequency selective mode only)
3. Input meter showing Peak and RMS values
4. Meters: TPK (True Peak), PK (Peak), RMS
5. Limiter gain reduction meter
6. Envelope follower settings (Attk, Rel etc.)
7. Compressor gain reduction meter
8. Non-linear transfer curve and signal metering
9. Output meter showing Peak and RMS values

Update from DS1-MK3 (v3.0) to DS1-MK3 plug-in

- New limiter Types added (Type 1 and Type 2, see description in the Limiter section)
- Processing resolution has been increased to 32 bit 192 khz (40 bit internal resolution)
- Added Waveform Display view for real-time gain reduction monitoring
- Options dialogs have been redesigned for better ergonomics in computer environment
- Preset management can now be done in the DAW or a dedicated Preset Collection tool
- Screen is now interactive. Users can click and change on various screen elements and settings (see the detailed description in Screen section)
- Original Data/Gain knob now performs only Gain change function

Limiter

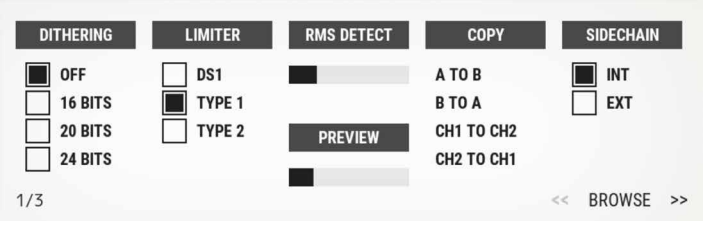
There are three types of limiters available in DS1-MK3 plug-in:

- Original Safe limiter from DS1-MK3 hardware
- New generation Weiss brick-wall Limiter Type 1
- New generation Weiss brick-wall Limiter Type 2 (True Peak)

Limiters Type 1 and Type 2 are designed to provide the most transparent sound for mastering applications performing at high RMS levels.

TIP: Use Type 1 for highest RMS values and Type 2 when True Peak limiting in needed.

Options Menu



Dithering Sets dithering to 16, 20 and 24 bits

Limiter Option Allows to choose the limiter types. (see Limiter section)

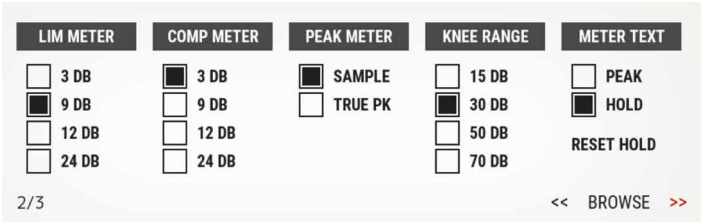
RMS Detect In the DS1-MK3 the peak measurement is supplemented with an RMS measurement with variable averaging time. The left endstop of this parameter switches the detection to Peak mode (default one).

Copy option allows to copy channels and settings from A to B, B to A, and channel settings from Channel 1 to Channel 2 and vice versa.

TIP: you may copy or switch the settings from the bottom panel of plug-in as well.

Sidechain Sets the internal or external sidechain.

Limiter meter range Changes the range of the meter.



TIP: It is possible to change the meters range from the main screen by clicking and dragging the meters.

Compressor meter range Changes the range of the meter.

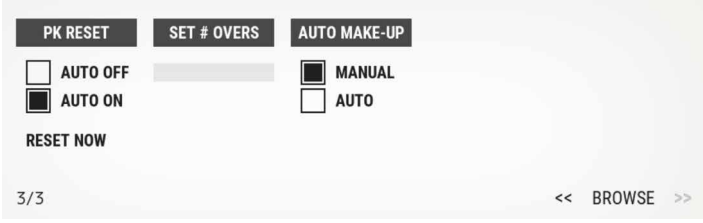
Peak Meter changes the meter type from on-sample peak meter to True Peak (intersample) meter.

Knee Range changes the visible range of the Knee on the main screen.

Meter Text parameter changes the peak meter from Peak to Hold. In Peak mode the meters read the instant values thus change values faster, while in Hold mode the meters update the peaks slower.

Reset hold resets the meter to the current peak value.

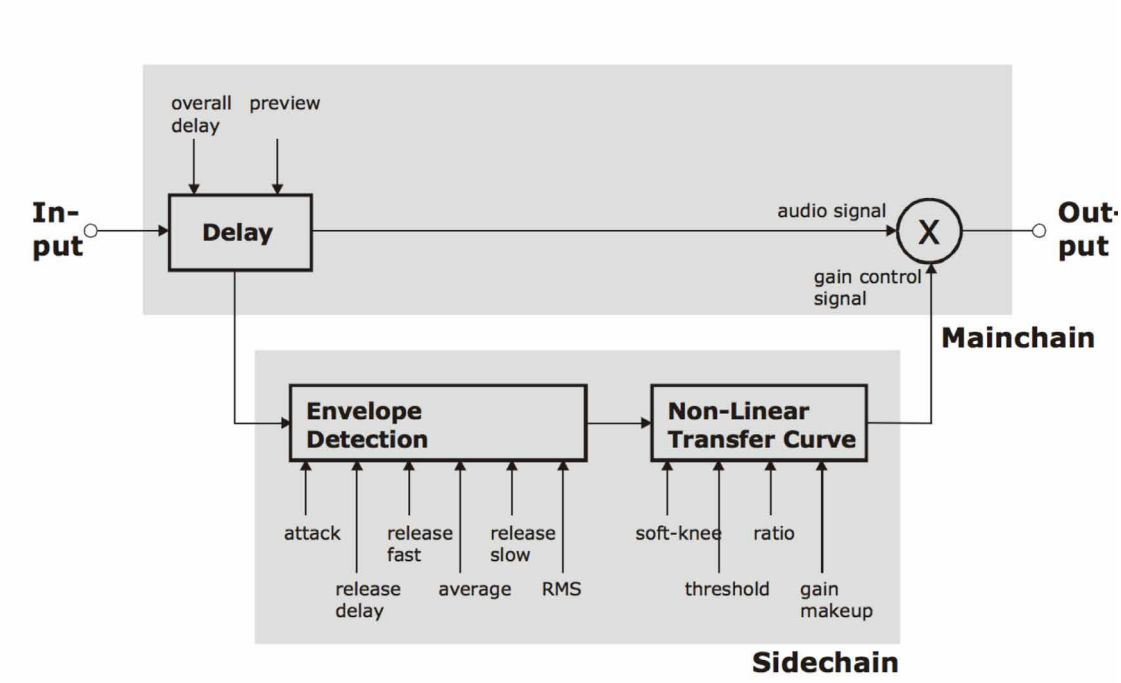
Peak reset sets the automatic peak reset option to ON and FF



Set # of Overs Sets the number on consecutive on-sample overloads to detect and report overloads. By default the number is set to 3.

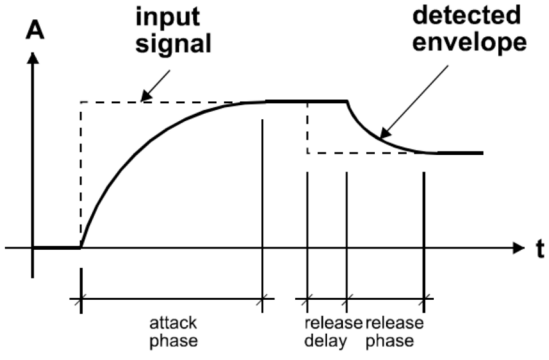
Compressor Parameters

Following is a detailed description of all parameters that can be adjusted in the compressor stage. Guidelines are given for settings (see also the factory presets), but optimal settings are usually programme dependent, so experimenting and listening is inevitable for satisfying results. **Graphic 1** shows schematic and parameters of the DS1-MK3 compressor stage:



Graphic 1: Compressor schematic.

Attack and Preview The longer the attack time, the more will the leading edge of fast transients pass by the gain reduction circuit unaltered (as in **Graphic 2** during attack phase). Very fast attack time settings such as 20µs, i.e. one sample period, do catch every transient, but may distort low frequencies. To utilise longer attack times and still catch fast transients, use preview.

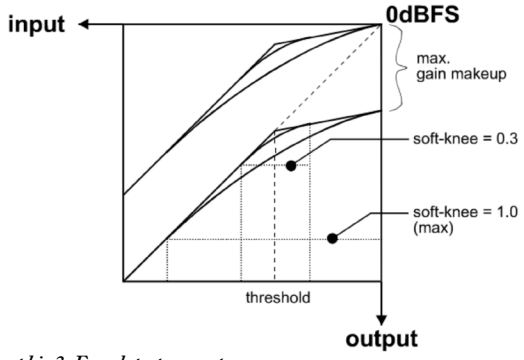


Graphic 2: Envelope parameters.

Release and Average The input signal is monitored with two different methods: peak amplitude and RMS value. The peak amplitude is the programme portion we don't perceive as very loud, however which can easily cause overloads. The RMS value of the programme material we perceive as loudness. Its variations contribute most to the dynamic range of the audio. The sidechain processor compares the two measurements it obtained from the peaks and the RMS. The ratio between the two determines which time constant would be the appropriate release, e.g. after short duration peak the faster release will be applied during the release phase. The time period over which the RMS value of the programme material is averaged is set by the "average" parameter. The effect of the average parameter can best be studied on the gain reduction meter. Fast "average" settings will cause most of the gain reduction meter to move very fast (depending on the "release fast" setting) , slower settings will just have the top part of the gain reduction move fast, with the bottom part depending on the "release slow" setting.

Release Delay Normally the release phase begins immediately after the programme material has fallen in level. The release delay determines how long the DS1-MK3 holds the current level before entering release (see **Graphic 2**). The resulting dynamic characteristic is very smooth and avoids otherwise typical 'pumping' effects. The ideal setting is programme dependent and experimentation may be necessary to achieve the best combination of all dynamic parameters.

Non-Linear Transfer Curve This part of the sidechain is responsible for the actual gain reduction calculations. The detected signal envelope is compared with the transfer curve, the gain control signal (see **Graphic 1**) is then adjusted accordingly if the envelope overshoots the transfer curve.



Graphic 3: Envelope parameters.

Threshold Sets the level which the envelope signal has to reach before gain reduction is applied. Ratio: Sets the slope of the segment above the threshold.

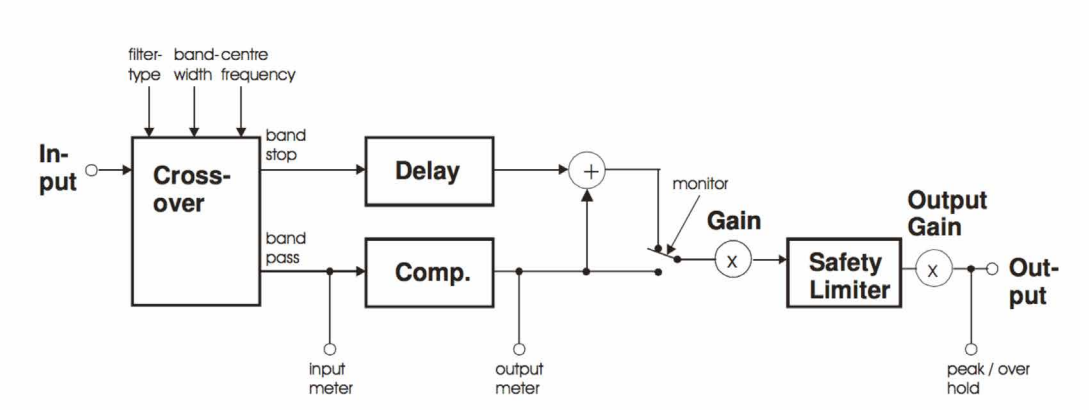
Ratio The Ratio can be adjusted for compression or expansion, depending on the requirement. For full band expansion (e.g. to reintroduce dynamics into an over-compressed signal), one will usually work with the gain makeup set to maximum. This means that a 0.0dB input will be passed through without gain, signals below the threshold will be reduced with a constant gain (depending on threshold and ratio setting, can be read from the display when touching the gain makeup controller), and signals between the threshold and 0.0dBFS get expanded. For full band expansion, one will typically have quite high thresholds (-1dB ... – 4dB), because of the resulting overall gain. The expander can also be used in frequency selective mode, for example to repair the effect of a misused multi-band compressor – for this purpose, the gain makeup can be set higher than the calculated maximum setting, to compensate for the overall gain that the expander causes. This can potentially cause overs (for loud signals in the expanded band), it is therefore good practice to engage the safety limiter.

Soft-knee Determines how much the knee is rounded. The maximum setting (1.0) chooses a curve that reaches from 0dBFS down to twice the threshold value. This implies that if soft-knee is engaged the signal is already affected even if its envelope lies below the threshold.

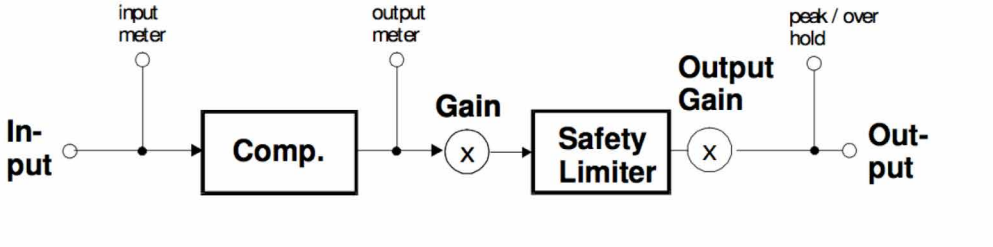
Gain makeup Moves the whole transfer curve up or down. In de-essing mode the gain makeup is usually set between the minimum (- 10dB gain) and "off" (0dB gain) to actually attenuate the processed signal, while in compressor (full band) mode the transfer curve usually gets to lie anywhere between "off" and "max", according to the amount of loudness correction one wants to apply. Once "max." is on, the transfer curve is attached to 0dBFS for maximum possible gain. "max" is engaged as soon as the curve reaches 0dBFS by manipulating any combination of the above parameters, or by selecting "max" from the gain makeup menu. This mode can be exited by manually reducing the gain makeup or by selecting "off" from the gain makeup menu.

Band Selective Mode vs. Full Range Mode

The DS1-MK3 can be used in conjunction with a frequency crossover (band selective mode) or straight (full range mode).



Graphic 4: Band selective mode.



Graphic 5: Full range mode.

Band Selective Mode In this mode, a frequency crossover is put into the signal path, splitting it up into two or three bands, depending on the filter type setting. One of the bands is compressed, the other is delayed by the same amount used as"delay" in the compressor stage. This mode is completely transparent if no gain reduction is applied.

Filter Type selects the available filter types: Low Pass, Band Pass and High Pass.

Frequency changes the center frequency of the selected filter.

Bandwidth selects the width of the selected frequency band.

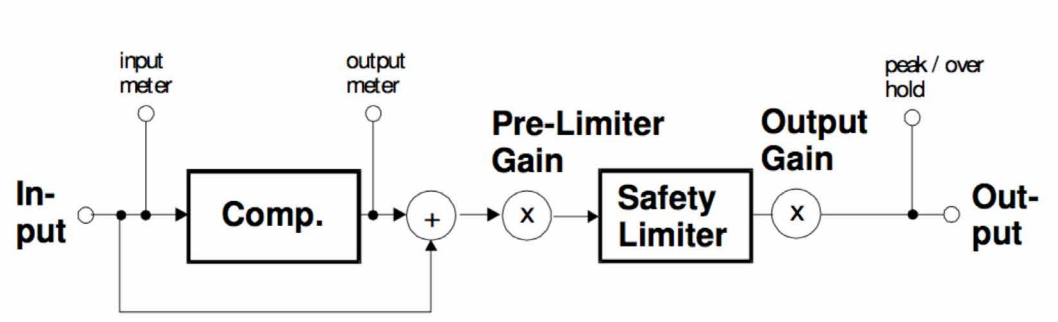
Parallel Compressor The DS1-MK3 can be used in parallel compression mode, where the compressed signal is added to the straight signal.

Parallel Full Band Compression As can be seen in **Graphic 6**, the input signal and the compressed signal are summed. The gain make-up value can be used to control the amount of compressed signal. The pre-limiter gain can be used to scale the mix before limiting.

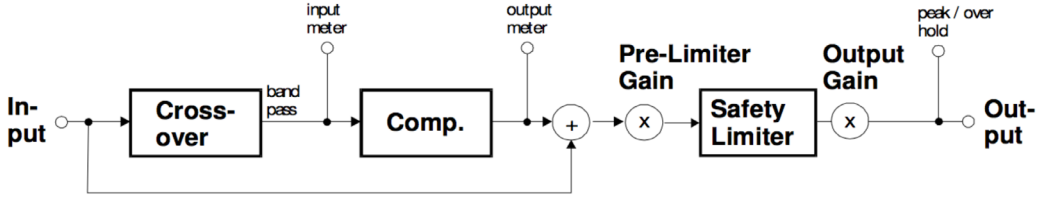
Parallel Band Selective Compression To add even more subtle compression, the compressed signal can be filtered before compression. Using the monitor function, one can listen to the filtered signal for optimal crossover setup (See **Graphic 7**).

Why parallel compression?

Read this quote from Bob Katz: "You want a patch that compresses without stomping on the attacks at all, preserves transients but gives you great inner detail? ... You have one control, one control only, and that is the output level of the compressor [gain makeup], the 'sidechain'.

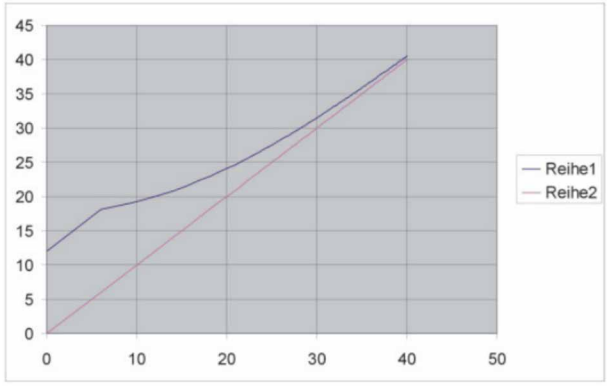


Graphic 6: Parallel band full compression.



Graphic 7: Parallel band selective compression.

You can bring out inner details without losing breathing, and hardly affecting the upper dynamics at all. Works great. Not just great, absolutely fantastically. Fattens bass



Graphic 7

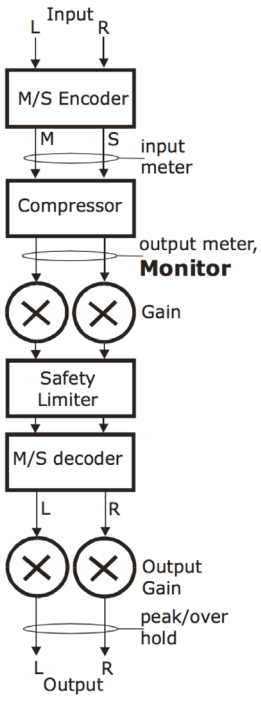
instruments without causing breathing or pumping. You name it, it works very well. Variations: Use it by split band [i.e. parallel] compression, so that (for example) high frequency inner details at low levels are enhanced". This last paragraph refers to parallel band selective compression (crossover in highpass mode).

Starting point for parallel compression: Set the threshold for -50 dBFS, ratio to 2.5:1, attack to the fastest (microseconds), release to mid, around 250-300 ms (adjust to taste if you hear breathing or pumping). The makeup gain governs the amount of compression.

Graphic 7 shows the effect of Parallel Compression on the transfer curve (upper graph). Lower levels are raised while higher levels are not much affected.

Ganged /Unganged

The DS1-MK3 allows for independent parameter settings in channels 1 and 2. If the ganged key is lit, both channels operate with the same parameter set. If the ganged key is not lit, the two channels can operate with different parameters. The CH1 and CH2, Mid/Side key assigns the knobs and the display to the appropriate channel.



Graphic 8

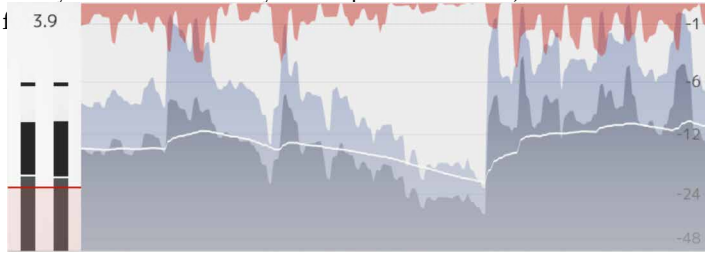
Mid/Side Mode and Sidechain Link

The DS1-MK3 can be switched to Mid / Side mode. In M/S mode it usually is preferable to set the channels unganged and the sidechain unlinked.

Graphic 8 shows the block diagram of the DS1- MK3 when in M/S mode.

Monitor Key

In band selective mode pressing the "monitor" key allows you to listen directly to the processed band, ideal



the effect of the compressor stage. See above for the Monitor output position. The Monitor output works as follows: DS1-MK3 in ganged mode: The stereo signal is fed forward to the output. If the band selective mode is selected, only the active band is fed to the output. DS1-MK3 in unganged mode: The currently selected channel (CH1 or CH2 or M or S) is fed to both L and R outputs. This allows for instance to listen to the S signal only in order to adjust the compressor optimally.

Waveform View

The Waveform view shows several meters in time: gain reduction, RMS and peak values.

Factory Presets

The DS1-MK3 has easily accessible factory presets for a quick setup which can then be refined and adapted to the programme being processed. The first few factory presets we created at Weiss Engineering, then a whole bunch of presets created by Bob Katz follow. Bob's presets come with a detailed on-screen description. Note that most of Bob's presets need a threshold and/or a ratio dialed in in order to work. We invite users of the DS1-MK3 to supply their favourite presets for adding them to the list. To access the factory presets press the Presets at the bottom of the plug-in.

In the Setup menu you can enable "Bob Ludwig mode" that shows current parameter values under each knob.

Credits

Paul Shyrinskykh – product design and product management. **Arvid Rosén** – DSP and framework programming. **Björn Rödseth, Kim Larsson** – framework programming. **Niklas Odelholm** – graphic design. **Ulf Ekelöf** – 3D rendering. **Daniel Delviken** - marketing. **Maxus Widarson** – quality assurance. **Igor Miná** – user manual layout.

68 Other Platforms

Reason Rack Extensions

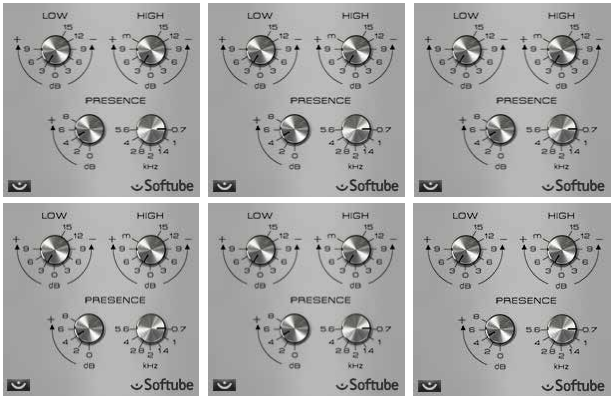
A lot of Softube products are also available as *Rack Extensions* (RE) for Propellerhead's host Reason. The sound quality and features are the same as in other plug-in formats. Rack Extensions are only sold through the Propellerhead web store.

The Rack Extensions licenses are completely separate from our VST/AU/RTAS/AAX licenses, are sold separately, and cannot be transferred.

For more information about Rack Extensions, please visit www.propellerheads.se.

Products Currently Available

Tube-Tech Classic Channel, Trident A-Range, Saturation Knob, FET Compressor, TSAR-1 Reverb, TSAR-1R Reverb, Valley People Dyna-mite, Spring Reverb, and Tube Delay are currently available in the Rack Extensions format.



Cakewalk Sonar ProChannel

ProChannel is an intramix interface for VST plug-ins, with easy accessible knobs directly in the mixer. ProChannel compatibility is included in the normal VST/AU license, and no separate installer is needed.

The ProChannel versions of the FET Compressor and Focusing Equalizer do not include all parameters, but the full VST versions are installed at the same time as

the ProChannel modules, and you can easily access them from the inserts menu.

Products Currently Available

The products included in the Mix Bundle: TSAR-1R Reverb, Passive Equalizer, Focusing Equalizer, Active Equalizer and FET Compressor are available with a ProChannel compatibility. Saturation Knob is included in Sonar X1 Producer Expanded.

UAD-2 Powered Plug-Ins

All Amp Room plug-ins (Vintage, Metal and Bass Amp Room), Tonelux Tilt and Tilt Live, Valley People Dynamite, Summit Audio TLA-100A, Tube-Tech CL 1B, PE 1C and ME 1B are currently available for the UAD-2 Powered Plug-Ins platform and Apollo High-Resolution Audio Interface. Marshall Legends, including Plexi Super Lead 1959, Silver Jubilee 2555, Bluesbreaker 1962 and JMP 2203 are available exclusively for UAD.

TC Electronic PowerCore

Vintage Amp Room has previously been available for PowerCore, but is no longer sold or supported.

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Brian Paul
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dhbaird@gmail.com
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